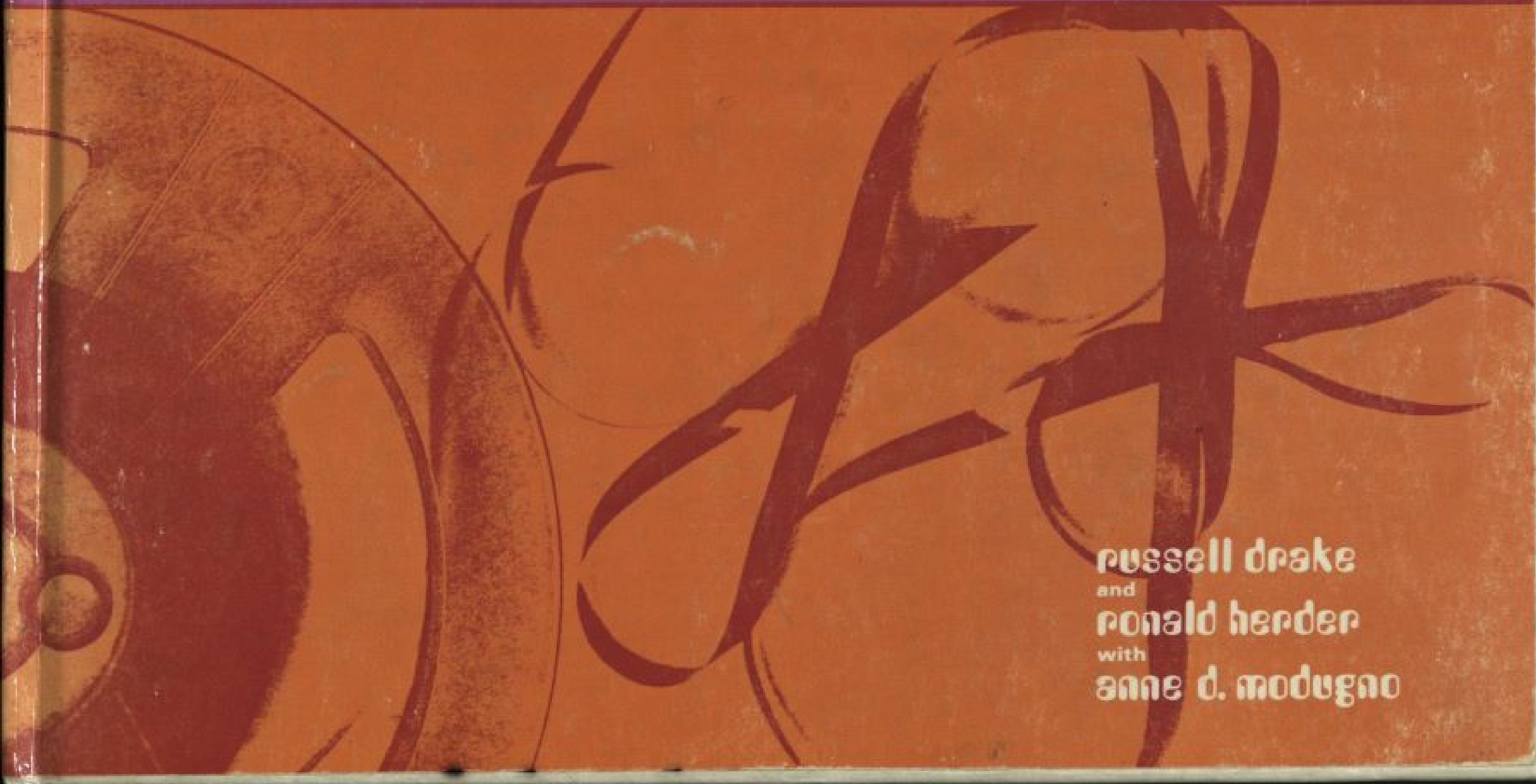
how to make electronic music



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how to make electronic music

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contents

introduction

Almost everybody is involved with music in some way—most people as listeners, many others as instrumentalists or singers. But very few people think of themselves as composers.

Composing, in past centuries, required years of specialized study in the elements of music, its form, its instruments, and its notation on paper. Composers who continue to write in this tradition today still need this training.

But in the twentieth century a new kind of music has had its beginnings—electronic music. This new idiom does not require the same specialized knowledge. It allows room for amateur activity. Anyone who can learn to manipulate a tape recorder can begin to enter this new world of sound transformation and can participate in creating music. That's what this book is about.

This new music is truly new in a very basic sense. It uses sounds that only a few short years ago had never been heard. The modern composer now has the choice of composing with these sounds alone or in combination with the whole body of familiar music—from folk songs, to pop, to classical. Before we get into the new processes, let's take a look into the past to see how music arrived at its present point of development.

In pre-electronic days a composer had to note on paper the nature, quality, and volume of the sounds to be translated by performers into a particular piece of music.

For over three hundred years, the materials for creating music were very much the same: the same musical scales, the same body of instruments played in the same way and in similar combinations. And while each great composer left his essential imprint on music, and the work of each generation of composers reflected what had gone before, they all nevertheless worked quite closely to the same basic rules of composition.

But as the nineteenth century ended, composers began to experiment with new sources of sound and new methods of presenting it. Some introduced "material" sounds into compositions, everyday sounds to augment the sounds of conventional instruments. Tchaikovsky scored a cannon shot into his *1812 Overture.* Another Russian composer called for the projection of colored lights to accompany his music (the first light/sound show composition) and the score listed the colors to be used. There were numerous other experiments.

Early in the twentieth century the French composer Edgard Varèse began a systematic exploration of the possibilities of

using the *total* world of sound in composition. In 1923 his composition *Hyperprism*, scored for sixteen conventional instruments, plus percussion, a siren, and a "lion's roar," was first performed in New York. The *New York Times* critic dismissed it as "the noise of a menagerie or of a factory accident."

Varèse's next composition a few years later was scored for an ensemble of thirteen percussionists playing thirty-seven different instruments—sirens, drums of various sizes, cymbals, wooden drums, anvils, a piano, and bells. Ionisation

Other composers followed suit—one with a work for ten pianos, electric instruments, and machines, including airplane propellers. Another included sirens and typewriters in his score. Still another, in a ballet suite, called only for fifteen percussionists.

Around 1925 a major event occurred when electric or, more correctly, electronic amplification and recording of sound were introduced.

With audio amplifiers, a host of new sound possibilities became available to the composer. For starters—composers were no longer limited to the built-in maximum loudness of an instrument or an instrumental combination. By means of direct amplification either in the concert hall or in recording, a piccolo could be made to sound louder than a trumpet, a violin could cut across a brass ensemble. The smallest triangle could drown out a full orchestra. for

The most important development of all came around 1950—the tape recorder. With it came such an incredible roster of new capabilities that now, a quarter of a century later, the boundaries of its possibilities are yet to be established.

The composer could now record sounds any time, any place, and—in a studio or workshop—manipulate, modify, mix, and control them to a minute degree never before possible.

Only five years later the RCA Synthesizer was developed. For the first time, all the parameters of sound—pitch, timbre, loudness, succession, duration, and the ability to shape sound into separate events—were brought under the control of the composer. A more elaborate version of the synthesizer, the Mark II, was developed in 1959 and was donated to the Columbia-Princeton Electronic Music Center at Columbia University. It cost two million dollars to build.

As composers began increasingly to work in this new medium, the need for smaller and less expensive synthesizers became more pressing. Robert A. Moog, working in consultation with various composers, was the first to develop a series of small, portable, and relatively inexpensive synthesizers. A host of others followed.

With the capabilities of the synthesizer added to those of the tape recorder, the definition of music began to expand. Now, not only could the composer utilize and modify sounds from nature and from instruments, with electronic generators, a whole new spectrum of sounds could be created and synthesized. Besides generating and synthesizing sounds, the synthesizer could also be used as a performance instrument, but this was not its most important feature.

In fact, in the development of electronic music, performers can be dispensed with entirely. The composer, recording and manipulating sounds on tape, is the performer and the tape itself the performance. The composition is an audio entity instead of notes on a music score: a concrete realization of Marshall McLuhan's idea—the medium is the message.

Now in classrooms throughout the country, students—many of them with no previous experience in music—are taking part in composing projects. You, too, can do it. All you need is a tape recorder, some relatively simple skills, some curiosity and imagination. In the following sections, you will be taken step by step into the exciting process of making music.

March 1975

Samuel P. Puner
President
Educational Audio Visual Inc.

sound transformation

Electronic music is created with a tape recorder by transforming raw sound into an organized expressive composition. Three phases are involved:

- collection
- modification
- composition

Raw sound can be collected from any source, such as nature, machinery, or musical instruments. To record these sounds you must be familiar with the standard operation of a tape recorder. If you are not, you can acquire this knowledge in an hour or two by studying your tape recorder manual or the instructions for basic tape recorder operations in the Technical Information section of this book.

Some additional recording skills are introduced in the Recording Experiments (pp. 10-13), which provide practice in the techniques of recording sound for the specific purpose of electronic music manipulation.

After sounds are recorded, they can be modified by any of the tape manipulation techniques described briefly below. These techniques perform two functions: they drastically extend the range of expressive possibilities of the original sound, and they disguise, or even destroy, its identity.

A sound can be used in its original state if you like the effect it creates. Usually, however, the listener's attention will be shattered by the sudden recognition of a sound: "Oh, that's a frog" or, "That's a rubber band." Tape manipulation techniques transform recognizable sounds into sheer audio sensation.

The next step is to combine these manipulated sounds into a structured statement or composition. This process is introduced in the Tape Experiments (pp. 35-38) and developed in the Composing Projects section.

Collecting, modifying, and composing, of course, do not follow a rigid, phase by phase, progression. Successful sound alteration experiments may suggest new sounds to be collected, or a partially finished composition may require remodification of existing material or the collection of still more new sounds to complete it. The first step is to understand the possibilities of modifying sound.

basic tape manipulation techniques

(Detailed instructions for each technique may be found on pp. 19-33.)

speed change

Sound can be either speeded up—and raised in pitch—or slowed down—and lowered in pitch—by recording it at one speed and playing it back (and possibly re-recording it) at another. This is one of the simplest and most useful techniques. It can produce extreme sound alterations.

backward sound

Sound can be recorded, the tape reversed, and the sound played backwards. This reverses the sound's envelope (its attack, sustain, and decay), producing forms of sound never heard before the invention of recording.

tape loop

The end of a sound can be spliced to its beginning, forming an endlessly repeating loop. Loops are valuable for ostinato effects. Several loops can be used simultaneously to play different rhythms against each other.

hand manipulation

Turning the reels or simply pulling a length of tape through the recorder by hand during playback provides speed variations unobtainable by using the fixed speeds available on the speed selector. Hand manipulation during recording—in addition to varying speed—can be used to chop the original sound into pieces simply by stopping and starting the tape.

echo

Sound can be echoed by utilizing an inherent time delay factor between playback and re-recording. Anything from a single repeat to slowly fading multiple echoes can be produced.

combining sounds

In addition to the possibility of recording separate live sounds simultaneously, a sound can be added to a pre-recorded one on the same track (sound-on-sound) or, using stereo or quad, on separate tracks (sound-with-sound). Combining sounds which by themselves may not be particularly interesting frequently causes them to interact or contrast with each other to produce a striking new entity.

spatial manipulation

Sound can be moved back and forth or positioned anywhere within a stereo or quadraphonic field. Spatial manipulation is an exciting aspect of composition which, with the possibilities provided by the electronic medium, can now be fully exploited.

feedback

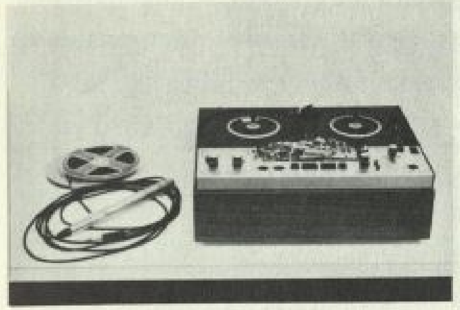
Acoustic feedback—microphone howl—can be used as a sound source. It can be recorded and modified by tape techniques like any other sound. Feedback can also be used as a modification technique. A sound can be fed back through a loudspeaker while it is being re-recorded to produce a range of interesting timbre (tone color) modifications.

editing

Sounds can be arranged and rearranged by cutting and splicing the tape. Editing can be used for composition to structure a succession of sounds, as well as for alteration to rearrange and recombine parts of sounds. For example, the attack of a gong can be cut off and spliced to the decay of a peacock's cry.

combining techniques

Though a single technique may alter the original sound beyond recognition, combinations of techniques extend the possibilities of transformation infinitely. The gong/peacock sound might be looped, slowed down, chopped up by hand-manipulated re-recording, then combined with a speeded-up Trinidad steel drum with echo added and a looped bullfrog played backwards. The possibilities are limitless.



equipment requirements

Sound manipulation is made possible by certain built-in features of tape recorders. In order for you to evaluate the sound-altering possibilities of your equipment, these features will be examined briefly.

reel-to-reel vs. cassette recorders

Only reel-to-reel recorders have the features that make sound alteration possible. Reel-to-reel recorders operate at more than one speed. Since the tape is exposed, it can be cut and spliced for editing, it can be manipulated by hand, and it can be made to by-pass certain parts of the machine in order to produce results not intended by the manufacturer.

Cassette recorders, in contrast, use encased tape and are usually single speed. They can be used in electronic music only for collecting sounds which must then be re-recorded onto reel-to-reel tape for manipulation.

Even for collecting sounds, reel-to-reel portables are preferable since they generally produce superior sound quality and make re-recording for manipulation unnecessary.

Since cassette recorders have only limited value in electronic music, the remainder of this section will deal only with reel-to-reel recorders and the differences among them which can affect tape manipulation techniques.

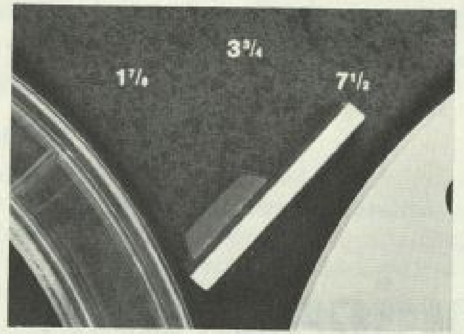
differences in reel-to-reel recorders

For producing electronic music there is no difference between using a tape recorder and a tape deck. (A tape deck is a component of an audio system and requires an external amplifier and speakers. A tape recorder is a complete system having its own amplifier and speakers.) The term tape recorder is used in this book to indicate either one.

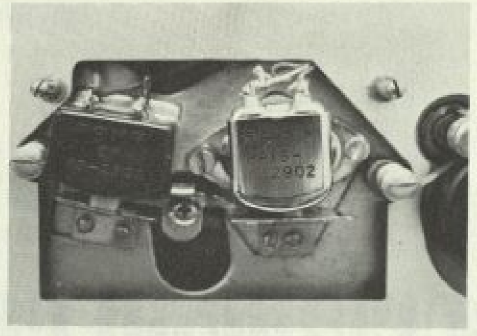
separate record and playback machines

Though a number of tape techniques can be performed with a single tape recorder, some require two: one to play back while the other re-records. If you have access to only one machine, you can probably borrow an extra one for the occasions when you will need two.

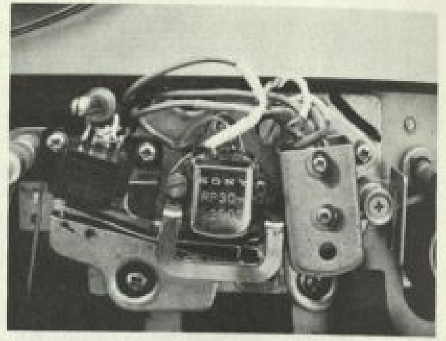
The best solution is to work as a group and pool resources



speed selector



two-head recorder



three-head recorder

(money, recorders, and hi-fi equipment). In the early days, this was the way some of the major tape studios were started. Film makers form cooperatives for the same reason: a group can accomplish things that are impossible for an individual—and have more fun doing it.

mono and stereo

Most sound alteration techniques can be accomplished on a mono recorder, but a few—like sound-with-sound and spatial manipulation—require stereo.

stereo and quad

Quadraphonic offers a few more possibilities than stereo: two additional channels of sound coming from separate speakers offer more possibilities for placing and moving sounds. Other advantages of two additional channels are the smaller number of generations of re-recording required to build up multiple layers of sound and the individual volume control over each channel which is useful in re-recording. The difference in cost between stereo and quad, however, makes non-essential advantages extremely expensive.

two- and three-speed recorders

Tape recorders have variable tape speeds. Some have two speeds, others have three speeds. Although speed change is one of the most frequently used techniques for altering sound, the difference between having two or three speeds available is not very significant. The possibility of repeated re-recording offered by a second machine makes an infinite range of speed change available even with two-speed equipment.

two- and three-head recorders

Three-head tape recorders have a separate head for each function: erase, record, and playback. Two-head recorders have an erase head, but use a single head for record and playback. This difference is significant. It affects both general recording and specific tape manipulation possibilities.

First, the combination record and playback head in a two-head recorder represents a technical compromise which produces a compromise in sound quality. The sound quality of a three-head recorder is superior.

Second, a separate playback head allows the machine to record and play back simultaneously. This allows you to compare the incoming sound with the recorded sound *while the recording is being made* (A- and B-testing). You can react to the actual sound being recorded and consequently have better control over the recording process than is possible by just watching the needle on the VU meter.

The ability to play back and record simultaneously produces a third difference—one which affects sound alteration. The distance the tape has to travel from record to playback head produces a split-second time lapse which can be used to create an echo: the sound played back a fraction of a second after recording is fed back to the record head and re-recorded as an echo of itself. This echo again reaches the playback head a fraction of a second later and is again re-recorded, etc., until the signal loses strength and the echo dies away. Since the time taken by the tape to pass from record to playback head produces the time lapse between echoes, varying the tape speed will vary the rate of echo. A three-speed recorder will produce three different rates of echo, while a two-speed recorder will produce only two.

half-track and quarter-track

The track is the area of the tape on which sound is recorded. Tape recorders designed for home use usually use a quarter of the tape width to record each channel of sound: a mono records one quarter at a time, stereo records two quarters at a time, and quad records four at a time. In the case of mono and stereo, the purpose of recording on only a quarter rather than the full width of the tape is so that the tape can be reversed and recorded again in the opposite direction. In the case of stereo, this doubles the amount of recording time per reel of tape, and, in the case of mono, it quadruples it.

Tape economy is obtained, however, at the expense of two factors related to the production of electronic music:

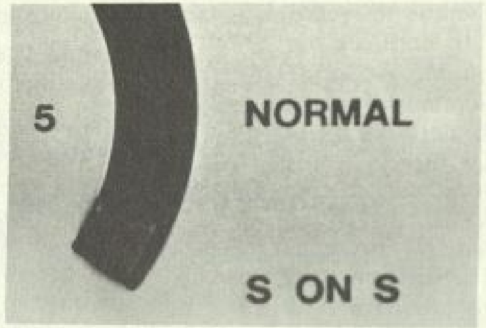
- The first is that track width affects sound quality: the wider the track, the better the sound.
- The second is that narrow track formats were designed to enable the tape to be recorded in both directions without erasing the first recording. It is therefore impossible to reverse the tape and play the recording backwards; the track will not match the position of the playback magnet. Backward sound is difficult to produce on quarter-track machines.

Other formats exist: half-track mono and stereo, and full-track mono.

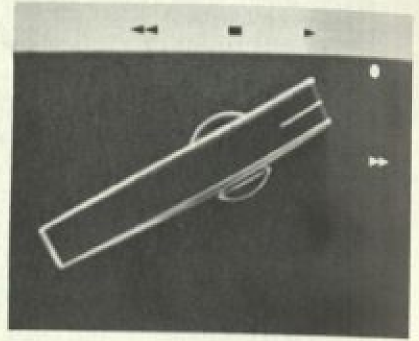
- Half-track mono, which records half of the tape width at a time, is a format which is also used in amateur recording. Though the sound quality is better than quarter-track, it has the same problem of preventing the simple playback of backward sound.
- Half-track stereo records one channel of sound on each half of the tape simultaneously. It is the standard format for professional stereo recording, but it is occasionally available—sometimes offered as an option—on home recorders. It not only produces better sound than quarter-track, but a recording made on one channel can be played backwards on the other channel by simply reversing the tape.
- Full-track mono, a professional format, is rarely used in amateur equipment. Since the sound is recorded on the full width of the tape, the sound quality is superior and the tape can simply be reversed and played backwards.

Sound *can* be played backwards on quarter-track recorders, but it requires techniques which are either tedious or produce an appreciable loss in sound quality. For this reason, as well as to obtain generally better sound, use wider track equipment if it is available.

mic/line mixing



sound-on-sound switch



control knob on pause

mic/line mixing

Some tape recorders can mix microphone signals with line signals (signals from another tape recorder, electronic instrument, etc.) and record them both on the same track—a form of sound-on-sound. This feature is extremely useful, but as other ways can be found to accomplish the same result, it is not essential.

built-in sound-on-sound and echo

Some tape recorders have built-in circuitry for echo, and—if they have mic/line mixing—may have built-in circuitry for sound-on-sound. A flip of the switch activates the circuit. However, the connections for either of these techniques can be made externally in a few seconds.

instant stop

Tape recorders usually have a switch or lever for 'instant stop' or 'pause control'. This is useful in locating sounds to be edited. If the control actually starts and stops the tape instantaneously, it is even more useful as it allows the precise recording of a succession of sounds on a single tape with little or no cutting and splicing.

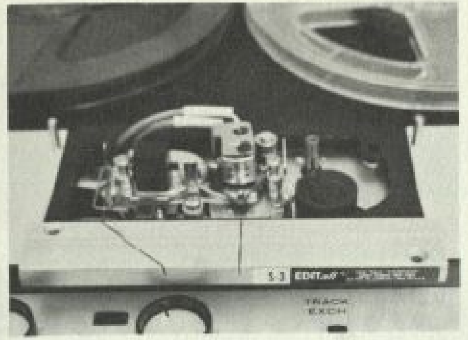
microphones

The microphone offered as an accessory for home tape recorders often provides lower sound quality than the recorder itself. A better microphone will usually produce a significant improvement in the quality of recorded sound. Recording for electronic music usually involves focusing on a single sound to exclude other sounds. This is best accomplished by the use of a directional (cardioid) microphone (see p. 98).

tape

The generations of re-recording required to alter sound degrade the original signal and accumulate tape hiss. The use of low-noise/high-output tape, rather than standard tape, will minimize degeneration and tape hiss by enabling you to record without distortion at higher levels.

Signals recorded at high levels, however, tend to print through to the next layer of tape on the reel. When played back, print-through sounds like a faint pre-echo of the original. Tape of



splicing block

1½ mil thickness, rather than thin extended play tapes, will reduce or eliminate print-through.

Storing the tape tails out—as it is after it has been played and not rewound—will cause any printed-through echo that occurs to *follow* the sound as a natural echo does, rather than to precede it. Tapes stored tails out simply require rewinding before, rather than after, they are played.

If your recorder is not biased for low-noise tape, it can be re-biased for a few dollars.

miscellaneous equipment

Editing equipment is essential. It consists of a relatively inexpensive splicing block, a grease pencil, razor blades, and splicing tape.

Headphones are extremely useful. They enable you to monitor recording. There are two kinds: one fits tightly to seal out external sounds, the other has foam cushions for comfort and does not seal out external sounds. Monitoring live sound can be done with the first kind, but not with the second.

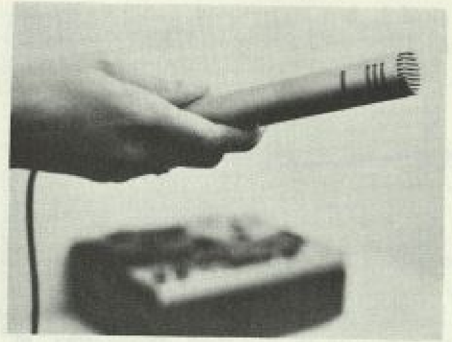
Microphone stands are useful, but something can usually be improvised in their absence: the microphone can be hung on or taped to a chair back, a camera tripod, a light stand, or a stepladder.

An assortment of patchcords and adapters are needed for connecting other recorders, record players, electronic instruments, etc., to your recorder.

A mixer is extremely useful. It provides separate level controls to regulate the proportion of each signal being combined, and combines them without distortion. Mixers, however, are expensive. An inexpensive alternative is to use "Y" connectors, although they may introduce a certain amount of distortion.

summary

- A reel-to-reel recorder is essential.
- A portable cassette recorder can be used to collect sound, but cassette recorders cannot modify sound.
- Most tape manipulation techniques can be performed on a two-head mono recorder.
- More techniques can be performed on a three-head mono recorder.
- Almost all of the techniques can be performed on a two-head stereo recorder.
- All of the techniques can be performed on a three-head stereo recorder (with the exception of the more complex spatial effects possible with quad).
- A second machine is frequently necessary—though it need only be capable of playback.
- The use of a better-than-average microphone is recommended.
- The use of low-noise/high-output tape is strongly recommended.



recording experiments

experiment 1

purpose

to distinguish between general noise and specific sounds

equipment needed

- tape recorder: battery operated portable preferably; if not available, either a tape recorder with long extension cords to reach outdoors, or a mic line long enough to reach outdoors
- microphone
- headphones

technique to be employed

basic recording

part 1, outdoors

1. Take the portable, or set up the recorder with extension cord, outdoors. If neither is possible, put the microphone outdoors, or at least hold it out the window. If you have a portable, find a reasonably quiet place to go. If you are in a busy city, do the experiment at night.
2. Plug in the mic, put on the headphones, set the machine on 'record', but set the pause control to prevent the tape from moving. (If you don't have a pause control you will be unable to monitor the sound without actually recording it. Since this is an experiment in listening, rather than in recording, use old tape and don't be concerned with the recording itself.) Turn up the volume and listen: motors, birds, dogs barking, wind, voices, airplanes, if you are in a rural or suburban area; if you are in an urban area: horns, brakes screeching, motors, sirens, air hammers, radios, voices, footsteps on sidewalks, etc. This barrage of sound is called ambient noise.

Normally you seldom notice ambient noise unless one of the sounds becomes loud enough to attract your attention. If it does, your ear is able to focus on this sound and more or less exclude the others. A microphone hears differently than your ear: it takes in everything equally. It will record all the sounds you are hearing. Tape some of the sound and listen.

part 2, indoors

ambient noise

Listen, as you did in Part 1, to various indoor environments. Tape some of what you hear. Can you find a place with no ambient noise? Keep places with low noise levels in mind as possible recording locations.

amplifier noise

Experiment with different volume settings while listening. Notice that at a certain point approaching the highest setting you begin to hear a hiss. This is amplifier noise in the recording circuitry. If you record sound using the range between this setting and maximum, you will record this amplifier noise along with it. When recording, avoid this range unless you are recording a steady loud sound that will mask it out.

microphone noise

With the volume at maximum, listen to the sound produced by your hand holding the microphone. If you have a reasonably sensitive mic, you won't be able to hold it without making noise. If you were recording a sound now, you would record this noise as well. That is why recording studios use mic stands. Study the way singers hand-hold mics in close-up shots on television so you will be able to hold one quietly.

conclusion

Recording sound with the clarity required for electronic music manipulation frequently requires high level settings which emphasize noise: ambient noise as well as noise from the recording process itself. Later experiments will show you ways to cope with this.

experiment 2

purpose

to distinguish the effects of different environments on recorded sound

equipment needed

- tape recorder with long extension cords
- battery operated portable (optional)
- microphone
- an assistant

technique to be employed

basic recording

part 1, indoors

1. Find a number of contrasting indoor environments:
   - a room with upholstered furniture, carpets, and drapes
   - a tiled bathroom
   - a clothes closet
   - a gymnasium
   - a long corridor
   - a swimming pool, etc.
2. Set up the recorder, ready to record, at one end of the room. One person will control the recording level while holding the microphone to record the voice of the other person. The other person will count or read a newspaper for a few seconds in a whisper, then in a normal voice, and then at the top of his voice, repeating the sequence at various distances from the microphone: directly into it, a few feet away, and at the other end of the room, modifying the procedure to fit the location (such as the closet) as necessary.

The person at the recorder can set the recording level for each distance change after listening to a brief sample of the voice but before starting the tape. The operator should strive for a maximum recording level without distortion.

part 2 (optional), outdoors

Repeat the same procedure used in Part 1, modified as necessary, in a variety of outdoor locations:

- an open lawn or field
- a narrow street or area enclosed by buildings
- a woods
- a tunnel or underpass, etc.

conclusion

Sound is obviously modified by the environment before reaching the microphone. Some of the sound waves pass directly to the microphone from the source. The rest, unless they are unconfined in an open space, strike environmental surfaces: soft surfaces absorb them, hard surfaces reflect them. Some of these reflected waves are picked up directly by the microphone while others bounce from surface to surface before reaching it. This reflection of sound is called reverberation.

A highly reflective—or acoustically active—environment can produce enough reverberation to overwhelm the original sound. A highly absorptive—or acoustically dead—environment can absorb enough sound to produce a dead sound.

Control of sound quality involves both a selection of the recording environment and a consideration of the distance between microphone and source. The farther the microphone is from the sound source, the more reverberation reaches it. The closer it is, the less reverberation reaches it.

You may also have noticed a similar relationship in regard to ambient noise. The farther the microphone is from the sound source, the more ambient noise is heard. The closer the microphone to the source, the less ambient noise is heard. The higher recording level required to pick up a distant sound also picks up more ambient noise. The lower level required to record a close-miked sound picks up less. This offers one solution to the problem of ambient noise control introduced in Experiment 1.

You rarely know exactly what you are going to do with a sound until after you experiment with it. The best strategy is to make a recording which will allow maximum latitude in sound manipulation possibilities. Use close-miking for a recording with minimum ambient noise and minimum reverberation—unless you want these elements mixed with your original sound for a specific purpose. You can always add reverberation to an already recorded tape by playing it back and re-recording it in a reverberant environment, but you can't remove reverberation from the original. You can mix in a track of ambient noise, but you can't remove it from the original.

experiment 3

purpose

to discover how differences in types and placements of microphones can alter sounds

equipment needed

- tape recorder
- microphones*: as many as you can find or borrow including, if possible, a carbon mic (A carbon mic is the kind of microphone used in telephones. It is inexpensive, with extremely low fidelity.)

*Check each one out first by plugging it into the recorder you are going to use and listening to what it picks up. If the sound is too faint, the microphone does not match your tape recorder. Don't use it.

technique to be employed

basic recording

part 1, variations in the way different microphones can "hear"

1. Set up the tape recorder in a reasonably quiet place with your collection of microphones ready to use.

One challenge of electronic music is to exploit all possible sound sources in your environment—even the most unlikely. Begin now with whatever is around you: knock things together, shake them, scratch them, or do whatever you have to do to them to get them to make sound (without ruining anything valuable).

2. Pick three to six contrasting sounds (ping, squawk, thud, etc.) that cover a frequency range from extremely low to extremely high, and try to record them as identically as possible with each microphone. Make a note of the order in which you use the microphones.

3. Listen to the tape. Do some mics noticeably drop out highs? Do others drop out lows? Do others drop out both? Do some exaggerate either highs, lows, or middles?

conclusion, part 1

The degree of sound alteration may vary from relatively slight to extreme. The carbon mic, if you used one, was probably the most extreme.

Though a mic with a wide frequency range and minimum distortion is more versatile, for specific purposes you might want to use one with a limited frequency range or one that produces tremendous distortion. Carbon mics are used as sound alteration devices in electronic music studios.

part 2, variations in the way a single microphone can "hear"

1. Select the most versatile microphone used in Part 1. Explore the sound sources of your surroundings by placing the microphone in a position that might alter the sound from the way you are accustomed to hearing it. For example, record a sound not normally considered loud with the microphone extremely close and the recording level high; or place the microphone inside the sound-producing object when possible.

2. Try using the mic as a contact mic: record your voice with the mic touching your throat or chest. Hard objects usually vibrate too much for direct contact, but you might experiment by taping the mic tightly to the object and then scratching or tapping the surface. (Be careful. Strong shocks can damage the mic.)

3. As an option, try the same thing outdoors. Experiment, for example, with the sound-producing possibilities of an automobile: the microphone can be placed in the carburetor air intake, in the exhaust pipe (if it's not hot enough to damage the mic); a burst of air out of the spare tire can be recorded, etc.

conclusion, part 2

A microphone is a creative tool. There is no *right* way to use it. The way you use it depends on the kind of sound you want to produce.

collecting sounds

The preceding experiments were intended to develop an awareness of key factors in collecting the kind of sound necessary for tape modification. These factors are:

- background noise
- amplifier noise
- noise from handling the microphone
- reverberation
- sound alteration produced by microphone placement, and possible sound alteration produced by the kind of microphone being used

Though the control over these factors is also important in standard recording, any problems they create are frequently emphasized when the sound is modified. If echo is added to a recording containing a dog barking in the background, for example, the dog bark will also echo. This will usually make its presence even more pronounced.

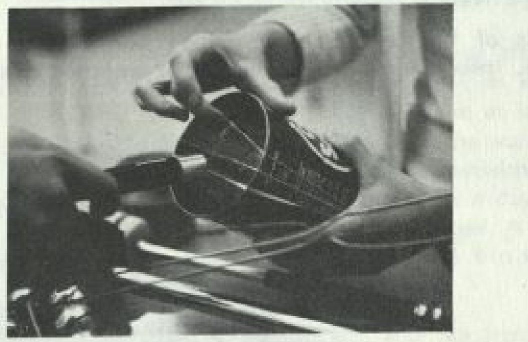
In Experiment 2, the technique of close-miking was seen to be effective in reducing both the level of background sound and reverberation. It is also effective in reducing noise from the recording amplifier, since a closer and consequently louder sound requires a relatively lower recording level. Amplifier noise is only a problem at highest recording levels.

Unless you deliberately want to exploit the characteristics of an inferior microphone, use the best one available (one with a wide frequency range and flat response). Normally you want to capture as much of the sound as you can. For example, when altering sound through speed change, high frequencies if slowed down sufficiently become middle frequencies; if you recorded no highs as a result of using a microphone with a limited frequency range, you could end up with a dull sound having only lows but no middles.

Several additional factors were mentioned in previous sections: recording at the highest possible level will result in a nearly saturated sound capable of surviving the degeneration and noise build-up resulting from modification and re-recording. The use of low-noise/high-output tape will permit you to record at a higher level without distortion, and will also produce a minimum of tape hiss.

These suggestions are meant to enable you to optimize your recording capabilities. Use them when practical. If at the moment they are not practical—if you can afford only standard tape, or if you don't have a very good microphone—don't wait until circumstances are ideal; simply proceed with the best you have. You can always upgrade your equipment later. If you can't get close enough to an interesting sound to close-mic it, try recording it anyway. It may turn out to be usable. If not, the experience will be useful in developing the ability to judge the latitudes of a successful recording.

Generally, the most useful sounds for modification are simple, clear, close-miked ones, recorded with a minimum of reverb and background sound. These will survive modification and combine well with each other. Complex or thick sounds do not survive modification without deteriorating into noises which will not combine well. A rubber band twanged close to the microphone, for example, will produce a more useful sound than traffic noise. A single guitar note (or simple interval like a fifth) will sound better slowed down than a guitar chord, which will become thick and noisy.



A number of useful sounds can be produced by common household objects: by a creaking door, by clanking tableware together, by running a wet finger around the rims of glassware, or running a thumbnail down the teeth of a comb; rhythms can be played on a typewriter; tunes can be played on a Touchtone telephone; there are numerous possibilities to exploit. The kitchen is a particularly rich source. Utensils can be made to yield various clangs, twangs, bongs, and squeaks. (Gongs from kettle lids slowed down can rival London's Big Ben.) Water sounds are available: a faucet dripping into a pan, water gurgling down the drain or splashed into a hot frying pan.

Stationery and hardware stores are good sources of inexpensive sound-producing objects: whistles and toy horns, rubber bands to twang, balloons to squawk, bells to ring, and boxes of things that rattle—like tacks, washers, and BBs. The sound of a Slinky toy, slowed down, is even "electronic."

Inexpensive musical instruments such as flutes, whistles, rattles, gongs, drums, kalimbas, and maracas, made primarily in Africa and Asia, can be found in import stores. A jew's-harp, available at music stores, produces a particularly rich sound when slowed down. Sounds from traditional musical instruments can be used and altered beyond recognition, or the instruments can be played in unconventional ways to produce unusual sounds. A piano, for example, can be made to yield a tremendous variety of sounds. A junk piano, which can be exploited mercilessly and without concern, is even better.

Equipment used in rock music can generate an interesting range of sounds: an electric guitar played by running a smooth object (a ballpoint pen or water glass) slowly up and down the high strings produces an "electronic" sound which is effective played back at slow speeds. A whole electronic piece can be made just by dangling a microphone in front of a guitar speaker-amplifier with the 'tremolo' switch on. The drummer's cymbals are a rich sonic source when played back at slow speeds. Even synthesizer effects can be produced—using almost any sound source—by the electronic modifiers used by many rock groups: wah-wah pedal, fuzz box, reverberation unit, phase shifter, envelope modifier, ring modulator, octave box, and so on.

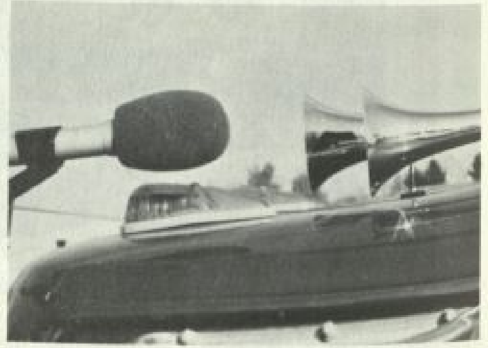
Junk yards can be explored. Metal pipes in various lengths and diameters can be used as percussion instruments. Small-diameter pipes can be played like bugles. Steel oil drums can be used as is, or modified in various ways, to produce percussive sounds. (The Trinidad steel drum played in Caribbean bands is an oil drum with the head hammered into tuned sections.) Automobile brake drums seem to have become almost standard instruments in the performance of contemporary percussion music.

Sound-effects records would seem to be a convenient source of material. The sound quality, however, is sometimes disappointing. Though some sounds may be usable, in general the recordings seem to be over-engineered to keep loud-speakers from buzzing or rumbling, and this tamed-down sound deteriorates quickly under attempts at modification. With even modest equipment you will probably find that you can make recordings yourself with sound that is richer and more saturated.

An exception to this is bird-call recordings. Since it is impossible to get close to a bird (except a caged one) with a microphone, professional bird recordings are usually made with expensive parabolic mics which focus distant sounds. These recordings are generally usable as source material. The relative clarity and high frequencies of bird-calls, unlike thick and over-engineered sound effects such as thunder and jet-plane take-offs, survive modification reasonably well.

Although exterior sounds generally are more difficult to record than interior ones—due to the problems of ambient noise and the frequent difficulty of close-miking—and although a number of them tend more toward sheer noise than usable sound, the exterior world is nevertheless a rich source of both natural and mechanical sound which cannot be duplicated in the studio. The suggestion to exploit all the sound-producing possibilities of an automobile, made in Experiment 3, can yield an array of source material which might be combined on the basis of its mechanical—rather than sonic—relationship. This suggestion, of course, could also be extended to a motorcycle—or, perhaps with more interesting results, to a tractor-trailer truck. If you know, or can befriend, a truck driver, you can record some unique sounds: the blast of the air horns close-miked, the percussive blast when the air brakes are released, and so on. All kinds of machinery, in both interior and exterior environments, can be explored in this way. A printing press, for example, in addition to producing a number of different sounds, is a source of complex and interesting rhythms. Rhythmic material is always a likely possibility for tape loops.

Sounds of nature present a unique contrast to mechanical sounds. Insects, unlike birds, can be approached close enough—if you move quietly—to permit a reasonably strong recording to be made. The complex, high-frequency sounds of crickets can easily be recorded this way. A microphone on the end of a pole can be held up into a tree to record a close-up of the rasping rhythms of katydids. Natural sounds, aside from producing interesting and unique material, are often more fun to record. Simply sitting quietly in the midst of the weird sounds and frantic activity occurring in an ordinary frog pond on an otherwise quiet spring night can be an unusual experience.



It is always fun to record—or to try to record—at the zoo, though you may not get many usable sounds. Animals in outside cages are usually too far away. The lion house at feeding time is a lively place but, since tiled interiors are the acoustical equivalent of swimming pools, the roars are overwhelmed by reverberation. An exterior cage for flying birds, on a day with few visitors, probably presents the most possibilities. A single peacock's cry, successfully recorded, can make the trip worthwhile.

Domestic caged birds are easier to record, and can frequently produce striking sound transformations. A crow—if you can find a tame one to close-mic—played back at quarter-speed, sounds like King Kong.

In collecting either interior or exterior sounds, record them at 7½ ips (or your highest tape speed). This will produce the best sound quality: the fastest speed will record the widest frequency range and produce the least amount of tape hiss.

After recording a sound at 7½ ips, play it back at slower speeds to hear the effects produced by slowing it down. Experiment with recording a few sounds at slower speeds, as well as at 7½ ips, and play them back faster to hear the effects of speeding them up. You will probably find that most sounds are more interesting slowed down than speeded up.

Material recorded at 7½ ips can always be speeded up later by re-recording it at slow speed onto another tape, and playing back the second tape faster. A little experience with the kinds of sounds that are interesting when speeded up can sometimes indicate when to make the initial recording at a slow speed in order to save this extra generation.

In order not to have to spend hours listening to tapes to find sounds you have collected, keep a recording log for each reel. It should contain the tape-counter number to be able to locate the position of the sound on the reel, an identification of the source, and any other information that might be useful: "0764—water dripping onto tray—rhythm interesting played back at 3¾ ips," etc.

Keep these tapes of collected sounds separate from the tapes of altered sounds which you will also begin to accumulate. Keep a similar recording log for each reel of altered sounds: "5603—jew's-harp from source reel #3—played back backwards at 1 7/8 ips—recorded at 7½ ips," etc.

As you collect sounds, experiment with the alteration techniques described in the following section. In the beginning, explore all the possibilities you can of altering each sound (or at least each different kind of sound) in order to develop the experience necessary to anticipate the particular technique, or

combination of techniques, that might be useful for a particular sound. A slowed down recording of a conventionally played guitar, for example, will simply sound like a slowed down guitar. A more radical alteration—perhaps re-recording it by hand-manipulating the tape during recording—may be required to produce something more interesting. Similarly, backward sounds frequently sound simply like something being played backwards. After the novelty wears off, they may not be very interesting. Some sounds, however, when played backwards, undergo a really significant transformation. A period of experimentation is required to develop a vocabulary of useful alteration techniques.

Unless you wish to alter the speed of a sound even further (by re-recording it at one speed in order to play it back at another) always record these alteration experiments—like the collected sounds—at 7½ ips. This will not only maintain the best sound quality, but will also simplify the process of editing: with the sounds farther apart on the tape, it will be easier to cut between them. These second and third generation tapes are the ones that will be edited. Even if you wish to use a sound without altering it, it should be re-recorded for editing in order to preserve your original material intact. You may want it for another purpose later.

Successful alteration experiments will frequently suggest other sounds that would be likely to produce interesting results. Your libraries of both collected and altered sounds will grow simultaneously. The more of these sounds you have accumulated, the more material and ideas you will have for compositions of real interest.

tape techniques

speed change

A difference between recording speed and playback speed will affect the tempo, pitch, and timbre of a sound.

If playback speed is *slower* than recording speed, the sound will have:

- slower tempo (one-half or less)
- lower pitch (one or more octaves)
- thicker or more resonant timbre
- longer playback
- slower attack, sustain, and decay
- more noticeable detail (including minute variations in pitch)

If playback speed is *faster* than recording speed, the sound will have:

- faster tempo (doubled or more)
- higher pitch (one or more octaves)
- thinner or brighter timbre
- shorter playback
- faster attack, sustain, and decay
- less noticeable detail

The degree of change—from moderate to extreme—depends on how many times the original speed is multiplied or reduced.

- Each time the playback speed is *halved,* the pitch is lowered one octave and playback will take twice as long.
- Each time the playback speed is *doubled,* the pitch is raised one octave and playback will take half as long.

The suggestion, made in the previous section, to record collected sounds at fastest speed and play them back slower—as well as to record occasionally at slower speeds for fast playback—will enable you to hear immediately the effect of a limited range of speed change.

With the use of a second tape recorder, and progressive stages of re-recording, the speed can be stepped up or down any number of times to produce an infinite range of speed change.

Tape recorders generally can play back better sound quality than they can record. To minimize sound loss when re-recording on a second machine, use the better of the two to record (the master), and the other to play back (the slave).

slow-down

1. Play back the original tape (recorded at 7½ ips) at 1 7/8 ips on the slave. (The sound is now two octaves lower.)
2. Re-record it on a second tape at 7½ ips on the master.
3. Place the second tape on the slave and play it back at 1 7/8 ips. (The sound is now four octaves lower.)
4. Re-record it on a third tape at 7½ ips on the master.

If the third tape is played back at 1 7/8 ips, the sound will be six octaves lower than the original. If the process is repeated once more, it will be eight octaves lower—by which time it will probably have become inaudible.

If your playback machine has only two speeds, you can step down only one octave—rather than two—at each stage of re-recording.

speed-up

At each stage of re-recording, play back the tape at the fastest speed and record at the slowest speed.

Again, a two-speed machine can step up only one octave, rather than two, at each stage.

A second machine is frequently needed to re-record speed-changed material to match other material with which it will be combined. A sound recorded at 7½ ips and played back at 1 7/8 ips, for example, requires re-recording at 7½ ips so that it can be spliced to other 7½ ips material.

Slow tape speeds produce a greater amount of tape hiss than fast speeds. Repeated re-recording using slow playback speeds will accumulate an appreciable amount of hiss. All of the suggestions previously mentioned concerning the use of low-noise tape, maximum recording levels, etc., should be employed to minimize this accumulation.

Speed change is easily the most useful sound-modification technique in the composer's repertory. Its effectiveness lies as much in its ability to modify timbre as it does in pitch and tempo modifications. Color change—especially valuable when applied to easily identified sounds—may completely mask the identity of the original source, frequently converting unpromising raw materials into interesting or even striking sound events.

backward sound

All sounds have a beginning (attack), a middle (sustain), and an end (decay)—three parts of a sound's contour or shape, called the envelope.

The most striking effect of playing a tape backwards—aside from the fact that the recorded events are heard in backward order—is that the envelope sequence is reversed. The sound begins with the end of its decay, moves backward through its sustain state, then ends with its attack. This reversal, completely foreign to our natural listening experience, produces a sound which invariably conceals its origin.

Methods for playing tapes backwards depend on the kind of tape recorder used.

Recorders with one-directional formats (full-track mono, half-track stereo, and quadraphonic) permit a recorded tape simply to be reversed and played back in the opposite direction.

In the case of full-track mono, reversing the tape makes no difference in the position of the recorded track since it occupies the full width of the tape. It can be played back in the same manner in either direction.

In the case of half-track stereo, the track—in an inverted position as a result of being reversed—can be played back on the opposite channel from which it was recorded.

In the case of quadraphonic, whatever channel matches the inverted position of the recorded track can be used to play it back.

With formats that were designed for recording in both directions (half-track and quarter-track mono, and quarter-track stereo) this possibility of simple playback in reverse has inadvertantly been foiled by the designers: the track in the inverted position will not match the position of the magnet in the playback head.

A recording made on half-track mono can be played back in reverse on a half-track stereo, quarter-track stereo, or on a quadraphonic machine, because there is a playback magnet that happens to correspond to the position of the inverted track. (Though this may be possible with other combinations of formats, avoid playing quarter-track tapes on half-track or full-track machines since the wider magnet will play back an additional track or tracks which—even though unrecorded—will add tape hiss.)

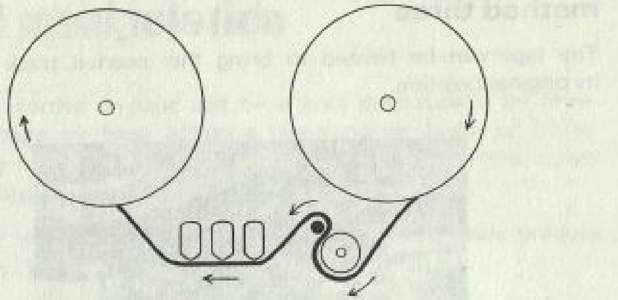
A tape *can* be played backwards on two-directional equipment if necessary. There are three methods:

method one

The tape can be threaded so that the transport mechanism automatically moves it backwards.

Place the tape, tails out, on the right spindle. Threading from right to left, pass it in front of the pinch-roller before threading it between the pinch-roller and capstan—and then behind the capstan and on through the normal tape path to the take-up reel on the left spindle.

The tape will move backwards using normal playback procedure—except that the take-up reel (on the left spindle) must be turned by hand.



This method accurately retains the sound quality of the original and is the simplest to do. Unfortunately, some tape recorders have obstacles which prevent the tape from being threaded in this manner.

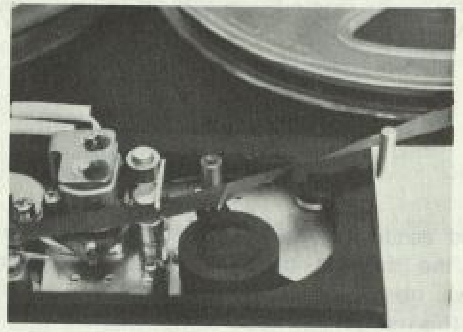
method two

The tape path can be temporarily raised so that it will correspond to the position of a playback magnet.

1. Place the tape, tails out, on the left spindle. Thread it normally through the transport mechanism—but remove it from the guide nearest the playback head, raise it slightly (away from the front panel of the recorder), and hold it in place with a pencil.
2. Start the tape in 'playback' and, while monitoring, use the pencil to move the tape slowly up and down on the heads until the track comes into alignment with a playback magnet and you can hear the reverse playback. Care must be exercised in keeping the track accurately aligned with the playback magnet to prevent the sound from dropping off. Due to the action of the tape guides and pressure pads, the tape will have a tendency to slip back into its normal path. Several attempts may be required to produce a successful dub.

method three

The tape can be twisted to bring the inverted track back to its original position.



Place the tape, tails out, on the left spindle and thread it normally. Make a half twist in it so that the recorded side is facing *away* from the playback head. Being both reversed and inside out, the inverted track will be back in its original position—but it will be played back from the wrong side of the tape. Since an appreciable sound loss will result, this method is recommended only as a last resort. If you have to use it, the sound quality can be improved slightly by dubbing the original at the highest possible recording level onto the thinnest extended-play tape available, and using the dub to play backwards—or the dub can be recorded at an even higher level on the wrong side and played back on the right side.

Although tape reversal may be applied to an entire composition, it is more frequently confined to smaller sections, or to individual sounds, for purposes of development, sound modification, and contrast. It is often more effective when used in combination with other techniques than when used alone: for example, sound may be reversed, slowed down, and have echo added.

tape loop

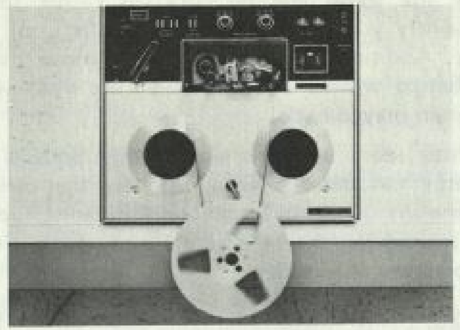
A tape loop is a circle of tape that runs continuously past the playback head, creating endless repetitions of a sound or series of sounds.

procedure

1. Locate and mark the beginning of the first sound. Cut the tape.
2. Locate and mark the end of the last sound (or the point at which you want the first sound to be repeated). Cut the tape.
3. Splice the end to the beginning to form a loop. (For splicing instructions, see pp. 95-97.)
4. Thread the loop in front of the heads and between the capstan and pinch-roller.

The first cut should be made at the very beginning of an attack so that a sudden change in sound will not be apparent. A noise-free splice is essential: the slightest click becomes noticeable during repetition.

Loops may vary in size from the minimum required to circle the playback mechanism, to a maximum which is determined



only by your ability to set up tape guides to keep it running in a path. It can circle the perimeter of the room, if you want, but a certain amount of tension must be maintained to keep it running smoothly past the playback head. Small loops can be held using a pencil or finger, larger ones by circling the tape around objects such as weighted jars or mic stands.

Loops are particularly valuable for ostinato effects. Due to the repetition of the events on a loop, their rhythmic relationship is emphasized. Once the loop is made, this relationship can be altered or adjusted by further editing: events can be rearranged, and spaces between them tightened or loosened by cutting out or adding tape.

The loop can be played back at different speeds—or in reverse—and re-recorded, making a range of variations of the looped events available for composition.

Two or more loops of the same or contrasting material, but of different sizes, played simultaneously will produce a texture of constantly changing rhythmic relationships.

Loops, or combinations of loops, may be used alone as compositional elements, or may be combined with other material to create multilayered textures of sound.

hand manipulation

Sounds recorded on tape can be altered dramatically by drawing the tape by hand across a live playback head—or by re-recording them while drawing the blank tape by hand across a live record head.

The two methods, though mechanically similar, can produce extremely dissimilar results.

Moving the tape by hand at varying speed during playback can produce extreme variations in pitch and tempo. The succession of original sounds, though, is continuous: if you play back "Mary Had a Little Lamb," all of the words—even though some may be reversed and repeated—will be present.

When moving the tape by hand while recording, the original sound—in addition to the possibility of extreme pitch and tempo variations—can be discontinuous: periodically stopping the movement of the tape (while the original continues on playback) chops up the sound by cutting out chunks of it.

The combination of chopping, with extreme pitch and tempo variations, produces the most dramatic sound alteration possible with any single tape technique.

hand manipulation during playback

1. Set up the tape for playback, with the machine on 'pause' or 'instant stop' in order to free the tape from the capstan.
2. Move the tape in either direction by hand-turning the reels—or simply hold the tape between thumb and forefinger and pull a length of it past the playback head. Experiment with speed variations.
3. If you produce an effect you want to preserve, re-record it on another recorder.

hand manipulation during recording

1. Set up the tape for playback on a slave machine. If the length of the selection is short, it can be looped to eliminate the necessity of frequent rewinding. (Live sound can also be modified with this method.)
2. Set up the master machine for recording, and press the 'pause' or 'instant stop' button to free the tape from the capstan.
3. While monitoring the source, begin recording either by hand-turning the reels, or by pulling lengths of tape through the machine by hand. Experiment with speed variations, and with stopping the tape periodically.
4. Play back the results.

An alternative is to by-pass the erase head on the recorder (see p. 28) to enable the tape to be moved in either direction without being erased.

When moving the tape by hand during playback, you can hear the results instantly, and you can easily control the degree of sound alteration: the faster you move the tape, the faster the tempo will be and the higher the pitch—and vice versa.

When moving the tape by hand while recording, you have to monitor the source rather than the tape (because when actually played back the speed will be constant rather than varied), and consequently you cannot hear the results until the tape is played back. Additionally, the faster you move the tape, the slower the tempo will be and the lower the pitch will be (and vice versa) when played back.

Though it may seem that this second technique is capable of producing only random results, a little practice can produce a surprising amount of control: while monitoring, you react to the sounds by moving the tape at a speed which is opposite to the effect you want to produce.

Hand-manipulation methods play an extremely valuable role in the repertory of tape techniques. Without them, we would be limited to the rigid pitch alterations produced by fixed speed changes. With the exception of expensive variable-speed tape recorders, they present the only possibilities of fluid pitch modification. They permit original sounds of otherwise fixed pitch to be modified in a range of effects from subtle variations to swooping glissandos.

echo

Echo can be produced on a three-head tape recorder: the time delay resulting from the distance the tape has to travel between the separate record and playback heads is used to produce echo by feeding the sound from the playback head back to the record head to be re-recorded as an echo of itself.

Echo can be added either to a live sound or to recorded material played back on another machine—or, a stereo recorder can play back pre-recorded material on one track, and add echo on the other.

The distance the tape travels from record to playback head can be extended—which consequently extends the delay time of the echo—by threading a single tape from the feed-reel of the recorder to the take-up reel of a separate playback machine.

A number of separate playback machines can be used to produce multiple echoes with differing time delays.

Echo can be made to alternate back and forth between stereo or quad channels.

Variables involving the use of more than one channel—as well as more than one machine—together with various patching possibilities and the use of different tape speeds, can be combined in countless ways to produce an unlimited number of echo configurations.

A description of basic methods follows.

The use of pre-recorded tape as a sound source enables you to add echo to already collected material. Any of the methods, however, can be used to add echo to live material by substituting a mic input for the line input indicated in the instructions. Using a live source will eliminate the need for the additional playback machine required in some methods.

The use of a direct-wire feedback circuit from playback to record is indicated. As an option, acoustic feedback can be used: the signal can be played back through a loudspeaker and picked up by a microphone for re-recording. Though this will produce the same echo configuration, some loss of sound quality is to be expected with any form of acoustic (rather than direct-wire) dubbing. The amount of loss will depend on the fidelity of the equipment.

The recording 'level control' is used to regulate the level of the original signal—but either form of feedback circuit requires a separate volume control to regulate the amount of echo which otherwise could overwhelm the recording. The playback volume control on the recorder is ideal for this purpose.

If your recorder has fixed—rather than volume controlled—output, the amount of echo can be regulated by the volume control on a mixer used to combine the signals. If a mixer is not available (requiring the use of a "Y" connector), the feedback circuit can be run through an amplifier to provide a means of control.

If a microphone is used (either to feed back the signal from the speaker, or to pick up a live source), a recorder with mic/line mixing capability will provide separate controls.

method one

mono recorder with source tape on separate playback machine

1. Combine the line output of the playback machine with the line output of the recorder (using a "Y" connector or mixer) and patch them to the line input of the recorder.
2. Set the recorder on 'record'.
   Adjust the recording level while monitoring the source.
   Set the monitor switch on 'tape'.
   Set the playback volume control at minimum.
3. Start the tape.
   While monitoring, carefully increase the playback volume control to produce the desired amount of echo. Be careful not to increase it too much or the echo build-up will quickly overload the recording.
4. Use different tape speeds to produce different echo spacings.

A speaker/mic feedback circuit can be used as an alternative to the direct-wire circuit. Connect the playback output of the recorder to a loudspeaker (through an amplifier if necessary). Connect a microphone to the mic output of the recorder, and position it to pick up the signal from the speaker. Use either mic/line mixing controls, or record and playback controls, to regulate the amount of echo.

method two

single stereo recorder to play source tape back on one channel and record echo on the other

1. Place the tape, with the source material recorded on track 1, on the recorder.

2. Combine the line output of channel A with the line output of channel B (using a "Y" connector or mixer), and patch them to the line input of channel B.
3. Set channel A on 'playback'.
   Set channel B on 'record', with monitor switch on 'source'.
4. Set the playback level of channel A at maximum.
   Set the recording level of channel B at an approximate setting.
   Set the playback volume of channel B at minimum.
5. Start the tape.
   While monitoring channel B, adjust the recording level.
   Change monitor switch to 'tape' and carefully increase the playback volume as in method one.
6. Use different tape speeds to produce different echo spacings.

A speaker/mic feedback circuit (from channel B playback to channel B record) can be used as described in method one.

method three

single tape threaded on two stereo machines in order to increase time delay from record to playback

1. Set up the playback machine to the right of the recorder. Thread the tape (pre-recorded on track 1) from the feed-reel of the recorder to the take-up reel of the playback machine. Thread it normally past the heads of both machines. By-pass the capstan of the recorder (see p. 95) to allow the tape to be pulled by the playback machine alone in the event of a slight difference in running speed.
2. Combine the line output of channel A of the recorder with the line output of channel B of the playback machine (using a "Y" connector or mixer), and patch them to the line input of channel B of the recorder.
3. Set channel A of the recorder on 'playback' (or the monitor switch on 'tape'), and set the playback volume at maximum.
   Set channel B of the playback machine on 'playback' (or the monitor switch on 'tape'), and set the playback volume at minimum.
   Set channel B of the recorder on 'record', with the level at an approximate setting, and the monitor switch on 'source'.
4. Start the tape.
   While monitoring, adjust the recording level and carefully increase the playback volume of channel B of the playback machine as in method one.
5. Experiment with different playback speeds, and with positioning the playback machine at different distances from the recorder, to vary the amount of playback delay.

This method can be used with mono rather than stereo machines but, unless recording from a live source, an additional playback machine will be required for the source tape.

A speaker/mic feedback circuit (from channel B of the playback machine to channel B of the recorder) can be used as described in method one.

The tape can be threaded from the recorder to any number of playback machines to produce multiple echoes with differing time delays.

method four

cross-echo using a stereo recorder with the source tape on a separate playback machine

1. Combine the line output of the playback machine with the line output of channel B of the recorder (using a "Y" connector or mixer), and patch them to the line input of channel A of the recorder.
   Patch the recorder's channel A output to its channel B input.
2. Set both channels of the recorder on 'record'.
   Adjust the recording level of channel A while monitoring the source.
   Set the recording level of channel B approximately.
   Set both channels on 'playback', or the monitor switches on 'tape'.
   Set the playback volume controls for both channels at minimum.
3. While monitoring both channels, start the tape and carefully increase both playback volume controls to produce the desired amount of echo.
   Adjust the recording level of channel B, if necessary.
4. Experiment with different tape speeds.

The echo will cross back and forth and be recorded alternately on each channel.

Speaker/mic feedback circuits can be used if they are separated acoustically to maintain channel separation—otherwise the effect will be destroyed.

The same cross-over patching configuration can be used with a tape threaded on two machines (as in method three) but—as both channels are recording—unless a live source is used, a third machine will be required to play back the source tape.

comments

Having the source material on one track and recording the echo on another (method two) enables the original to be played back at any volume while fading the echo in and out at will.

An interesting alternative to method two is simply to eliminate the feedback circuit and allow the source to be re-recorded on the other track as a single delayed repeat. If desired, the two tracks can be mixed down to one, and a third repeat added—and so on. The effect of limited repeats differs considerably from a dying-away echo.

Stereo echo, as well as repeat, can either be mixed down and played back in mono—or the channel separation can be maintained, producing an effective means of converting monaural sound to a form of stereo. The most spectacular of these conversions are the cross-echo configurations such as the basic one produced by method four.

Reverse echo can be produced by playing a source tape backwards, re-recording it with added echo, and then playing the re-recording backwards—which will reproduce the source material in its original sequence, but with the echo preceding it.

The result of the extreme time delay produced by the use of the same tape on two or more machines might more properly be considered a musical canon than an echo.

Although specific instructions are not given for quadraphonic recorders, any of the methods can be duplicated—or elaborated further to exploit the possibilities of echo on four channels. For example, channel 1 output can be patched to channel 2 input, channel 2 output to channel 3 input, channel 3 output to channel 4 input, and channel 4 output back to channel 1 input. Or, the output of any channel can be split and patched to the inputs of several channels, etc.

Although echo can be an exciting form of sound modification, it can become a bore if over-used. The ease with which it can be added to any sound makes this a definite danger.

combining sounds

Sounds can be combined to be heard simultaneously (as well as in succession through the process of editing) by the use of several techniques. A sound—live or pre-recorded—can be added to another pre-recorded sound on the same track, sound-on-sound, or on a separate track, sound-with-sound.

sound-on-sound

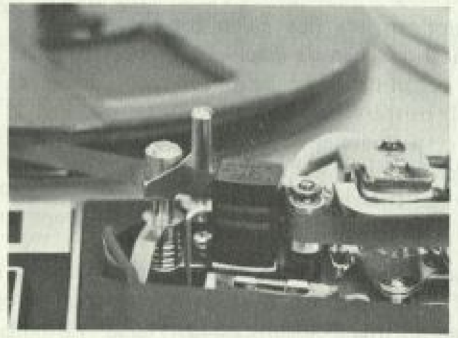
Using a three-head stereo tape recorder, a pre-recorded track can be played back on one channel and re-recorded on the other while additional material—either live or pre-recorded—is added.

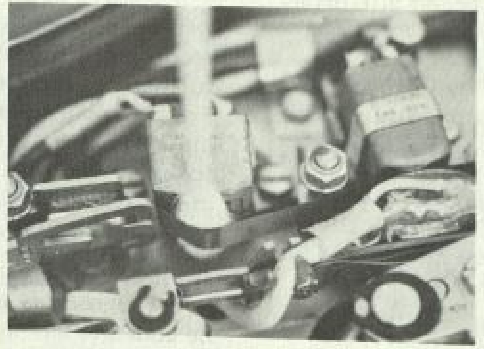
Some tape recorders have a sound-on-sound switch to activate internal circuitry which routes the signal from the output of the playback channel to the input of the recording channel. In the absence of this feature, simply patch from the line output of the playback channel to the line input of the recording channel.

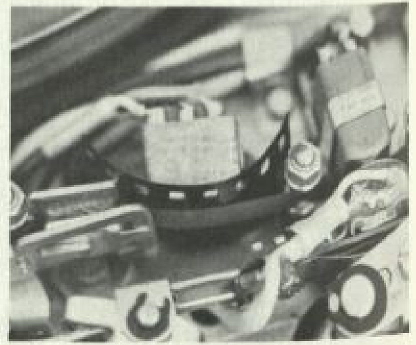
Mic/line mixing (or a separate mixer) can be used to add a live signal to a pre-recorded one. If the signal to be added is from a line source (another tape recorder, electronic instrument, etc.), it can be combined through the use of a mixer or "Y" connector. If a "Y" connector is used, playback volume controls will be needed to control the balance of the mix.

Using a stereo recorder for sound-on-sound employs the two channels as if they were separate mono playback and recording machines. Two mono machines can be used in essentially the same way to mix a live and a pre-recorded sound, or three machines can be used to mix two pre-recorded tapes.

There is another method for doing sound-on-sound on a mono recorder. If the action of the erase head (which erases the tape before each recording is made) is defeated in some way, a second recording can be made directly on top of the first one. In order to do this, the erase head can be by-passed by threading the tape behind it or, if the mechanics of the recorder prevent this, a cotton swab or piece of photographic film can be placed between the erase head and the tape to prevent contact. This method, unfortunately, produces a significant loss in sound quality and should be used only in the absence of other possibilities. Since the strength of the first recording will be considerably diminished when over-recorded by the second one, it should be recorded at maximum level and the second one added at a lower level so that they will come out more or less even.







sound-with-sound

Using a three-head stereo recorder, a pre-recorded track can simply be played back and monitored while a second recording is added on the other track.

The two recordings—unlike those of sound-on-sound—remain on separate tracks and can be played back in stereo. Or, if they are to be combined with other sounds, having the material separately tracked permits a final mix-down to be made with control over the proportion of each element.

The method, however, has a disadvantage. Since the playback of one channel is monitored while the recording is being made on the other, the two tracks will not be in exact synchronization as the record and playback heads are in different positions in relation to the tape. The method is useful only when exact sync is not required. When it is, use sound-on-sound which does combine material in exact sync since both programs are monitored from the same position as the source.

Two separate tracks can be recorded in sync by playing back a pre-recorded track on a separate machine and recording it on one track of a stereo recorder while live or pre-recorded material is added on the other track. Both inputs can then be monitored at the source.

tracking

The problem of inexact sync using sound-with-sound has been overcome in professional multi-track equipment by adding circuitry which allows the record head to function temporarily as a playback head. In this way the pre-recorded track, or tracks, can be monitored from the same position on the tape that the new track is recorded on—resulting in exact sync of all tracks. Playing back from a head designed to record produces an inferior sound quality, but it is adequate for monitoring.

This feature, called by various names such as Sel-sync and Simul-sync, is available on some four-channel recorders designed for home use.

Adding separate tracks in synchronization is more convenient than going through numerous stages of re-recording. Even more significant, however, is that it results in fewer tape generations—with consequently less loss in sound quality—and that it permits a final mix-down to be made with complete control over each element.

Although a four-track home recorder does not have the possibilities of a multi-track recording studio, groups of tracks can be mixed down in stages: after the first three are recorded, they can be mixed down to one—and then three more can be added. Or, after adding two more, they can again be mixed down to one so that two more can be added . . . and so on.

Tracking is useful, but not necessary, in producing tape music since the number of elements to be combined is usually not overwhelming. It is more useful in recording a synthesizer, which frequently requires more tracks.

spatial manipulation

Sounds can be positioned or moved in space laterally (from side to side) within a stereo field. They can be positioned or moved circularly (around the listener) in a quadraphonic field. They can also be positioned or moved radially (toward or away from the listener) in a mono, stereo, or quad field.

In stereo, lateral positioning depends simply on the relative volume of the channel producing the sound. If the left channel is producing sound at full volume while the right channel is at zero volume, the apparent source of the sound will be the left speaker. If the right channel is at full volume and the left channel at zero, the apparent source will be the right speaker. If both channels are producing at an equal volume, the apparent source will be midway between the two speakers. If one channel is producing a greater volume than the other, the apparent source will be somewhere between the midpoint and the speaker for that channel—the exact position being determined by the exact difference in amplitude.

If a number of sounds are recorded simultaneously, but each sound is recorded at a different proportion of amplitude on each channel, they will appear on playback to originate from different positions. At least five distinct positions can be perceived simultaneously: left, right, center, midway between center and left, and midway between center and right. Aside from the fact that the positioning of sound in space is an aspect of composition which can be interesting in itself, the individual sounds can be perceived with greater clarity than if mixed down to a single source.

A single sound can easily be positioned with a stereo recorder by splitting the source—either mic or line—with a "Y" connector, patching it to the input of both channels, and setting the recording level for each channel in the relationship required to produce the desired position.

Positioning more than one sound at a time requires—in the absence of expensive mixing equipment—that each sound be recorded on both tracks of individual stereo tapes, and then be mixed together by playing them back on separate machines using the playback volume controls to produce the correct amplitude proportions.

Moving sounds laterally in space, called panning, uses the same principles as positioning—except that the amplitude proportions are varied during recording. On playback, this produces the illusion of movement. If the recording level of the left channel is decreased from maximum to minimum, for example, while the level of the right channel is simultaneously increased from minimum to maximum, the sound will appear to move from the left speaker to the right.

Radial positioning (the apparent distance of a sound from the listener) is dependent on both amplitude and reverberation. A relatively loud sound with little or no reverberation will appear to originate from a source relatively close to the listener. A softer sound with more reverberation will appear to originate from a more distant source.

If two sounds are heard simultaneously, the louder one will generally seem nearer. Adding reverberation to the softer one will strengthen the effect.

Echo added at 7½ ips can be used to simulate reverberation.

Radial positioning is accomplished with relative ease with the use of a mixer, or by controlling the playback volume of the source tapes being combined in a mix, after having added echo to the sounds intended for distant positioning.

Radial *movement* is slightly more complicated. As a sound moves toward you, not only does the reverberation decrease with the apparent increase in amplitude, but the pitch increases slightly as well. As a sound moves away, the reverberation increases as the amplitude decreases, and the pitch decreases slightly. This pitch change, called the Doppler effect, can be observed, for example, in the whistle of a train going by: the pitch is higher as the train approaches than it is after it has passed.

Though these effects can be simulated more realistically with a synthesizer, they can be approximated—or, perhaps more interestingly, they can be exaggerated—with a tape recorder.

1. The sound selected for radial movement can be re-recorded, moving the tape by hand, to produce a slight bend in pitch—either up or down. This is the most difficult step and may require several recording attempts. Once a successful take has been recorded, it can be re-recorded in reverse for use in moving the sound in the opposite direction.

2. Echo added at 7½ ips can be used to simulate reverberation. Method two, described in the section on echo (p. 25), is ideal for this purpose: the sound (with the pitch bent upward) recorded on one track, can be faded in while the echo, recorded on the other track, is faded out—moving the sound closer to the listener. The sound (with the pitch bent downward) can be faded out, while the echo is faded in, to move the sound away into the distance.

The effect may be made more interesting by combining the radial movement with panning or, if using quad, the sound can be made to move toward the listener, pass through him, and fade away in the opposite direction.

All of the methods for positioning and movement can be used with quad—except that complex panning effects may require more than two hands.

feedback

Acoustic feedback—microphone howl—occurs when a microphone picks up sound from its own monitor speaker, producing a circular path for the sound to be endlessly fed from microphone to amplifier to speaker and back again. This continuous re-amplification of the sound produces maximum volume: a howl.

The distance between speaker and microphone determines the wavelength of the sound being passed through the circuit. Since high frequency sounds have short wavelengths, and low frequency sounds have longer wavelengths, the frequency of the howl can be varied by simply varying the distance. Generally, holding the microphone close to the speaker produces high-frequency feedback; holding it farther from the speaker produces low-frequency feedback. As the speaker/mic distance may sometimes pick up a segment of a longer wave, rather than a whole wave, some frequency variation can occur.

Feedback is the only sound that a system designed to reproduce sound can actually produce on its own. Like any other sound, feedback can be recorded and modified by tape techniques.

recording feedback

Set up the tape recorder for mic recording. Set the monitor switch on 'source', and connect the output to a speaker. With the amplifier gain at medium-low, and starting at a low recording level, hold the mic a few inches from the speaker and tap it with your finger to induce feedback. If it does not occur, keep increasing the recording level slightly. Gain and recording levels must be carefully controlled to prevent damage to the amplifier or recorder. When feedback occurs, start recording while experimentally varying the speaker/mic distance. The gain will have to be increased as the distance is increased.

A feedback duet can be composed on a stereo recorder by producing contrasting frequencies on the separate channels.

feedback modification

In addition to its use as a sound source, feedback can be used to modify other sounds.

1. Set up a stereo recorder (or master/slave mono recorders) for playback on one channel, and for recording on the other. Set the monitor switch for the playback channel on 'tape', and for the recording channel on 'source'.
2. Combine the outputs of both channels (using a mixer or "Y" connector) and connect them to a single speaker.
3. Set the output gain at medium-low, and set the recording level at medium-low.
4. Start the tape, pick up the played-back signal from the speaker with a mic, and carefully increase the mic-level to record the feedback. Experiment with different volume settings and with varying the distance between speaker and mic.

If the gain is carefully controlled, the events being played back can be fed back through the circuit without producing a howl. A range of interesting effects, from extreme distortion to subtle timbre modifications, can be produced.

editing

Editing is arranging or rearranging a sequence of sounds by cutting them apart and splicing them back together. Unwanted sounds or silences can be deleted, and the length of silences altered. The process consists of locating the precise parts of the sounds to be joined, marking them, and splicing them together. Directions will be found in the section on technical information, pp. 95-97.

A splice, whenever possible, should be made at the beginning of an attack. The sudden onset of transient sounds occurring in the attack will mask the abrupt transition from one event to another.

With the use of pause control or instant stop, cutting and splicing can be reduced—or sometimes eliminated—by start-stop editing.

1. Place the tape with the first sound to be dubbed on a slave. Locate the exact beginning of the attack and, using the pause control, set it for instant play.
2. Locate the exact point where the sound is to begin on the recording tape and, using the pause control, set it for instant record.
3. Start both tapes simultaneously. The click produced when the tapes start will be masked by the attack in the same way that a splice is masked.
4. Stop the recorder at the exact point where the sound ends. Then stop the slave and thread it up for the next sound.
5. Back up the recording tape an inch or two so that the click caused by stopping the tape will be erased or recorded over by the next sound. Repeat the process for each event in the sequence.

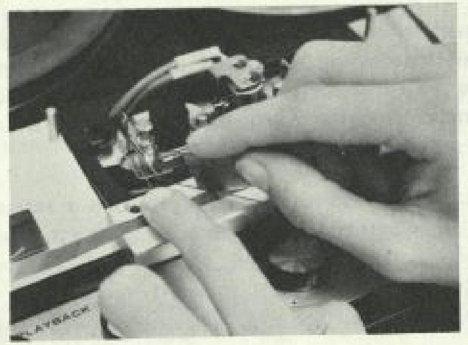
Audible clicks that result from recording sounds without sharp attacks—or even sections beginning with silence—can be removed by cutting and splicing.

Aspects of composition to be considered when combining sounds are discussed in the following section and in the section on composition, pp. 70-80.

The editing process can also be used to modify sound. The parts of a single sound—its attack, sustain, and decay—can be cut apart and recombined in various ways to form a new sonic event:

- The parts can simply be rearranged.
- One part can be reversed in relation to the others.
- A part from an altered recording of the same sound can be substituted. For example, a slowed-down attack can be substituted, or a sustain with echo.
- Parts of totally different sounds can be combined.

This micro-editing can be extremely tedious. It is not a process to be applied to an entire composition, but its potential for radical transformation makes its use on a limited scale worth considering. Record material to be modified at your fastest tape speed so that the sounds will be spread out on the tape in more manageable lengths.



tape experiments

A limited use of tape techniques can generate a variety of sounds. For example, one original sound of medium pitch, recorded at 7½ ips, can be played back at 3¾ ips and 1 7/8 ips to produce two additional sounds. If the original is re-recorded at 1 7/8 ips, playback at 3¾ ips and 7½ ips will produce two more, making a total of five sounds. Playing these five backwards will produce a total of ten sounds derived from one original.

Of the ten, those produced by speed change will contrast with each other in pitch and duration. They will be similar in envelope. They will vary in timbre—that is, they will have a degree of difference but, since they are derived from the same source, they will have a degree of similarity as well.

Those produced by reversal will contrast with the others in envelope, but be similar to them in pitch, duration, and timbre.

Any of the ten, through the use of different recording levels, can be made either to contrast or to be similar to another in volume.

If you want to create a sequence of sounds, a combination of these contrasting derivations is likely to be more interesting than merely repeating the original sound. Monotony is not satisfying.

At the same time, some aspect of commonality would establish a unifying relationship between the sounds. An unrelated hodgepodge is not satisfying. The ten sounds, however, having a degree of timbre similarity, already have a strong relationship.

If you succeed in combining these related sounds in a sequence which has an interesting degree of contrast, the repetition of the same degree of contrast would again become monotonous. Some kind of change (or contrast in the degree of contrast) is necessary to sustain interest.

The ten sounds may be arranged to produce a range of differing degrees of contrast. There is less contrast, for example, between a forward low-pitched sound and a reversed low-pitched sound than between a forward low-pitched sound and a reversed high-pitched sound. You might begin the sequence

using relatively subtle contrasts and progressively increase them to a maximum: a climax. Or you could produce an anticlimax by decreasing the contrasts from maximum to minimum. Or you could create a mirror structure that has maximum contrast at each end and minimum in the middle, or minimum contrast at each end and maximum in the middle.

experiment 1

purpose

to create a sequence of sound events derived from a single source

Using ten sounds as described above, arrange them in a climaxing sequence of at least twelve events. Use any or all of the ten, and repeat any as often as you like. Try to make the sequence as long as you can while keeping it pruned down to essentials by eliminating any elements which do not directly contribute to the effect.

Start-stop editing can be used to record a trial sequence, which can then be rearranged or altered by cutting and splicing.

experiment 2

purpose

to create a texture from a single sound source

1. Loop the sequence made in Experiment 1.
2. Play back the loop on a slave and record 60 seconds of it on channel A (or on a mono recorder if you don't have stereo) while hand manipulating the tape to vary the pitch and tempo. Try varying the recording level at the same time if you can.
3. Rewind and, using hand manipulation, record a contrasting track on channel B. (If using mono, by-pass the erase head and do sound-on-sound or, preferably, record from two playback machines.) Try to structure the degree of contrast between tracks in some form such as climax, anticlimax, or mirror.
4. Repeat steps 2 and 3, by erasing and re-recording, until you are satisfied with the results.
5. Play back the two tracks in stereo and then through a single speaker to hear the difference in effect.

(Frequently, mono playback can emphasize the relationship between simultaneous events. Compare the two forms of playback in all the following experiments with the exception of Experiment 6, which is designed for stereo.)

experiment 3

purpose

to create a texture from two sound sources

1. Using the same loop, repeat step 2 of Experiment 2.
2. Select an additional sound source to contrast with the initial one. Develop it into a sequence similar to that of Experiment 1.
3. Loop the sequence, and record it on channel B as in step 3 of Experiment 2.
4. Repeat steps 1 and 3 until you are satisfied with the relationship between tracks.

experiment 4

purpose

to create a texture from three sound sources

1. Mix down the two tracks recorded in Experiment 3 on channel A using a new tape.

2. Select another sound or sounds of contrasting or related timbre. Using start-stop editing, record them on channel B as individual events or groups of events in contrast to the fluid continuity of those on channel A. They might, for example, be short staccato sounds used as accents, to enliven dull places.

3. Keep erasing and re-recording the events on channel B until you are satisfied with the relationship between tracks.

A group of sounds, though organized into an interesting sequential and simultaneous form, cannot continue to hold the listener's interest indefinitely. At some point something new has to occur: another contrasting yet related group of events.

Popular songs frequently have a contrasting section, a bridge, to separate the main sections. The bridge, however, has to have some musical relationship to the main part—for example, similar rhythm patterns or similar phrases in a different key—or it would sound as if a totally different song had suddenly begun.

Tape music has the same requirements. A contrasting yet related section can be created by varying some elements of the original while keeping others constant. Duration and volume, for example, can be varied while pitch range and timbre are kept constant, so that essentially the same sounds will seem either to slow down or to speed up while becoming either louder or softer. Timbre and pitch can be varied while duration is kept constant, so that a new group of sounds retains the same tempo as the first. Having some element or elements in common will permit contrasting sections to be perceived as parts of a whole.

experiment 5

purpose

to structure two complementary sections of sound events derived from two sources

first section, first track: rhythmic loop (sound A)

1. Select from your collection—or record—a strongly rhythmic sequence of a single sound, such as water dripping into a pan or the sound produced by tapping a microphone with your finger.

2. Dub it on channel A at 7½ ips.

3. Rewind, and dub the same or a similar sequence on channel B at 3¾ ips, or mix them together using sound-on-sound.

4. Play back the two tracks (in mono) and select the most interesting playback speed.

5. Select a short section having the greatest rhythmic interest. Dub it and preserve the original.

6. Loop the dub. Edit, if desired, to produce a strong repetitive rhythm.

first section, second track: contrasting tones (sound B)

1. Record the loop on channel A for 60 seconds.

2. Select from your collection—or record—some long thick sounds, like kettle-lid gongs or piano tone-clusters. Play them back at slow speeds and choose the most interesting.

3. Record a sequence of the thick sounds—with optional use of echo and reversal—on channel B. The sequence may be recorded on a separate tape first for editing, or may be recorded directly onto the final tape using start-stop editing and erasure to develop a cohesive sequence. If the sequence lasts less than 60 seconds, the remainder of the loop recorded on channel A can be erased or cut off.

second section, first track: 'siren' loop (sound B)

1. Select one of the thick sounds. Re-record it, using hand manipulation to produce a glissando—a smooth transition from a low pitch to a high pitch, or vice versa. A little practice, or several attempts, may be required to produce a usable take.

2. Re-record the glissando in reverse.

3. Splice both glissandos together into a loop to produce a tone of continuously ascending and descending pitch, like a siren. Edit the splices, if necessary, to eliminate jumps in pitch.

4. Optionally, make two copies of each glissando before step 3. Make a second loop about 2/3 the size of the first so that the two can be played together in a constantly changing relationship.

second section, second track: random rhythm (sound A)

1. Record the siren loop (or loops) for 60 seconds on channel B, adding echo. (Echo added at this stage, rather than before looping, will mask the splices.) Leave a blank section of tape between the end of the last section and the beginning of this one. Fade out the ending.

2. Dub the tape produced in steps 1 to 3 of the first section, first track and splice in various short lengths of leader to separate the events into random groups. (The randomness should contrast with the repetitiveness of the loop of the same events.) The sequence can be structured by thinning out the groups to form an anticlimax, or by spacing them closer at the end to form a climax, etc.

3. Add the sequence of random groups to the siren track by dubbing it on channel A. If the sequence is too short in relation to the first track, the tape can be re-recorded to fade the siren out sooner.

4. Cut out the blank tape and splice the beginning of this section to the end of the previous section. The repetitive rhythm on channel A in section one will change to a random rhythm on channel B in section two. The sounds of fixed pitch on channel B in section one will change to a repetitive siren on channel A in section two. If the juncture between sections can be improved either by deleting material, or by adding silence, resplice it.

5. Play back the two sections. Was the device of using the same material, but alternating the modification techniques, successful in creating unified but contrasting sections?

A sequence of sounds may be arranged in a looser structure than, for example, an obvious climax, yet can still be interesting. A less obvious structure may sometimes be even more interesting. Sounds may be arranged in exciting combinations through experimentation without preconceived planning. Frequently, intuition is a better creative impulse to follow than logic. It can often produce a mess, however, which may require some kind of logical analysis to clear up. Intuition and logic need to coexist so that each functions when needed. In the next experiment, try to let intuition guide you as far as it can, but be ready to use some logical form of organization to pull things together whenever it seems necessary.

experiment 6

purpose

to structure a long series of events on the basis of experimentally derived relationships

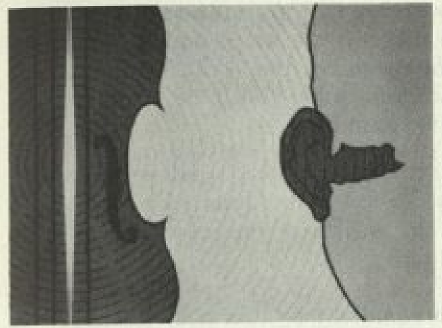
1. Set up to record acoustic feedback.
2. Record a number of feedback sequences, changing pitch by varying the speaker/mic distance.
3. Experiment with different playback speeds.
4. Edit some sequences to produce abrupt rather than gliding pitch changes.
5. Add echo to some sequences.
6. Experiment with re-recording them (including those with echo) using feedback as a modification technique.
7. Record contrasting sequences on either channel of a stereo recorder. Vary the recording levels. Use contrasting tape speeds.
8. Record a sequence on one channel and a feedback tone of fixed pitch on the other. Alternate channels.
9. Select a long sequence, or loop a short one, and record it with cross-echo on both channels.
10. Pan a sequence back and forth between channels. Use mic/line mixing to add staccato feedback squawks while panning.
11. Record a series of squawks on one track, rewind, and re-record them on the other to produce a delayed repeat. Or do this on a separate tape first using a slower tape speed to prolong the delay.
12. Explore as many ways as you can to modify and combine the material.
13. When you have exhausted all the possibilities, cut out the less interesting sections and experiment with rearranging the remaining ones.
14. Re-record the edited tape using feedback as a final timbre modification. Howls, or echo, may be added at this time to mask any transitions that seem too abrupt.

conclusion

Experiment continually in combining different sounds. Set up two recorders on playback and listen to different combinations of the tapes produced by your alteration experiments. Keep open to further modification possibilities which might produce a more interesting contrast or relationship between two sounds. When you find an interesting combination, record it.

Play back these combinations and look for a third sound to add to them. Explore ways of developing the combinations into longer sections. Analyze complete sections for possibilities of creating contrasting sections to add to them.

Continue collecting new sounds, modifying them, and combining them in this way. Look over the experiments in the synthesizer section (pp. 58-69) for possibilities of adapting them to tape techniques, and then go on to the composing projects beginning on p. 70.



acoustics

Acoustics is the study of the behavior of sound waves.

vibrations

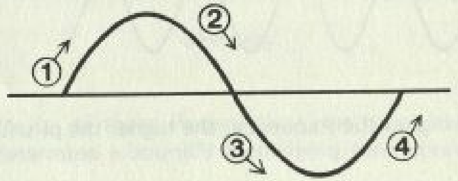
Every sound we hear is produced by vibrations. Just as waves of water ripple outward from a center of disturbance (a stone dropped in a still pond), air disturbances move outward from a vibrating body in the form of sound waves.

A clear and simple example of this is the plucked violin string, illustrated above. The waves move from the string, through the air, then to the ear's membrane, which vibrates in turn to carry the sound message to the nerve center.

waveforms

If we draw a horizontal line to represent any object at complete rest, we can then make a picture of the vibration in the form of a curved line.

When the object is disturbed into movement, it vibrates (1) outward in one direction, (2) back past the point of rest, (3) outward again in the opposite direction, then (4) back again toward the point of rest.



cycles

This vibration—pictured as one complete wave motion—is called a cycle.

If we use this one-cycle waveform as a starting point, we can explore the dynamics of sound—that is, how it moves, and how it can change its four basic characteristics: pitch, timbre, loudness, and duration.

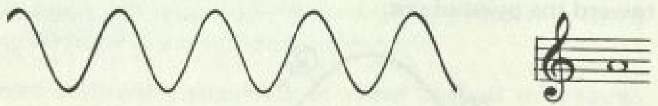
pitch and frequency

The pitch of a sound—how high or low it is—is determined by how many cycles occur in one second.

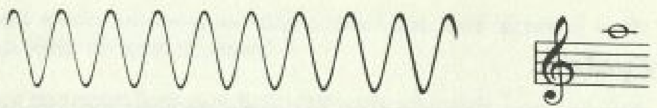
Although vibration speed has an enormous range—from less than one to many thousands of cycles in a single second—the audible range for the human ear is between approximately 20 cycles per second and 16,000 cycles per second. Vibrations below this range are called sub-audio; vibrations above are called ultrasonic.

The number of cycles per second produced by a vibrating body is called its frequency, a number expressed by the abbreviation 'cps' (for cycles per second) or, more recently, by 'Hz' (for hertz). "Tuning A," for example, is the pitch produced by a frequency of 440 cycles per second or 440 Hz.

- The lower the frequency, the lower the pitch.



- The higher the frequency, the higher the pitch.



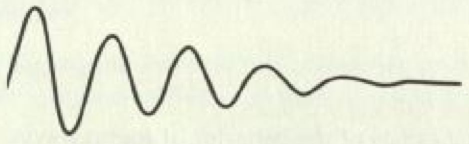
loudness and amplitude

The loudness or softness of a sound also depends on vibrations: not, like pitch, on how many occur in one second, but rather on the height of the wave's peaks.

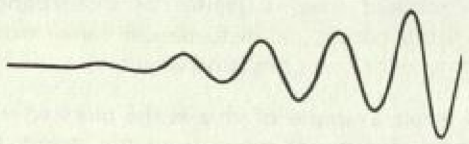
This characteristic—the waveform's amplitude—is determined by the strength of the force that sets the sounding body in motion. The stronger the controlling force, the greater the amplitude and the louder the sound; the weaker the force, the smaller the amplitude and the softer the sound.

Sounds (other than electronically produced ones) are in constant motion with continuous changes of volume from beginning to end. Here, for example, are waveform representations of three sounds with varying amplitudes:

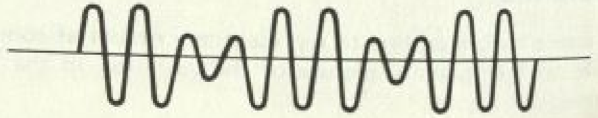
- a sound that begins loudly, then dies away



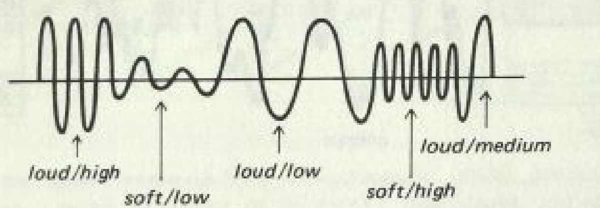
- a sound that begins softly, then becomes gradually louder



- a sound that is alternately loud and soft



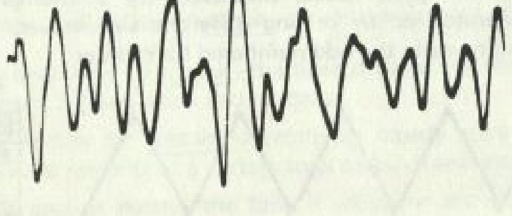
Note that the frequency of the sound remains constant despite changes in amplitude. If it did not—that is, if there were simultaneous changes in both frequency and amplitude—the waveform could be represented this way:



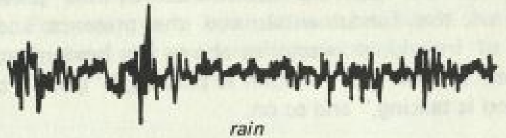
timbre and overtones

The S-curve waveform we've used to this point is a picture of a sound so simple and pure that it can be produced only by a tuning fork or by the electronic vibrations of an oscillator. The waveform is called a sine wave, and its sound—familiar as a radio or TV test tone—is called a sine tone.

But sounds in the world around us—a bird call, the human voice, the sound of any instrument—are complex structures with waveforms that look like this:



dog bark



rain

thunder

bird chirp

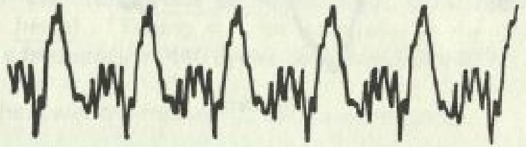
The complexity of these shapes—their particular curves, peaks, and dips—determine a sound's identifying tone color or timbre.

We can think of timbre as a parallel to the painter's colors: primary hues, mixtures (primary blends in varying proportions), and shades (degrees of brightness or darkness).

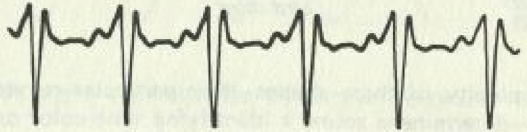
In the world of sound, the pure base color is called the fundamental; the added color tones are called overtones or partials. Overtones are further classified as harmonic—if they exist in a consistent mathematical ratio to each other—and non-harmonic if they do not. (Noise, and some complex timbres like bell sounds, are products of non-harmonic overtone structures.)

The timbres we hear are determined by the presence and strength of the fundamental, and the presence and relative strength of individual overtones above the fundamental. These factors tell the ear that "a violin is playing," "glass is crashing," "my friend is talking," and so on.

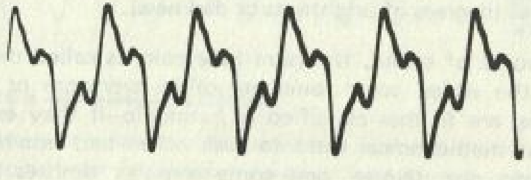
The following illustrations show the waveforms of three sound sources producing the same pitch at approximately the same degree of loudness. Note that while the frequency is identical and the amplitude similar, the waveforms—and consequently the timbres—are different.



violin

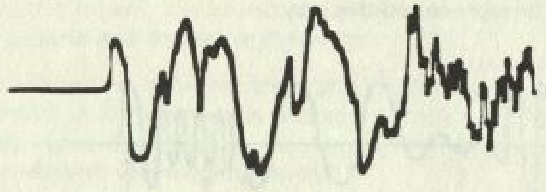


trumpet



clarinet

This illustration is an example of a sound structure consisting of non-harmonic overtones.

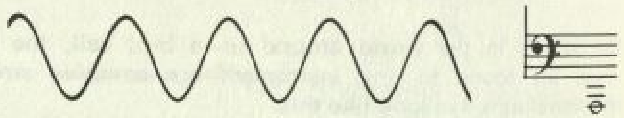


chimes

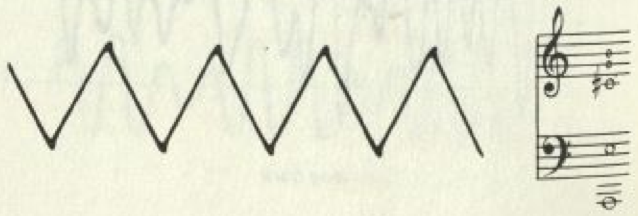
electronically produced waveforms

In the following section, we will be dealing with four basic waveforms which do not exist in nature. These are produced by the synthesizer's generators to make up the raw material for electronically produced and modified sound:

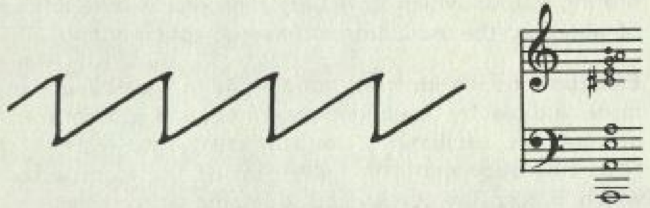
- the sine wave, produced by the sine-wave generator, and containing no overtones



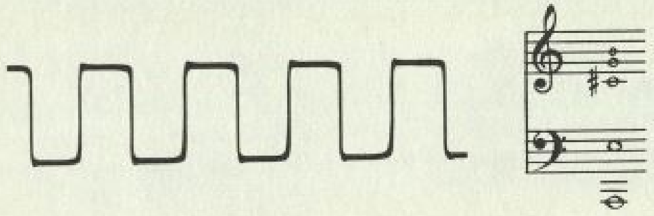
- the triangular wave, produced by a triangular wave generator or by mixing different sine waves, and containing only the odd-numbered harmonics



- the sawtooth wave, produced by a sawtooth generator or by mixing sine waves, and containing all harmonic overtones of the fundamental



- the square wave (a form of rectangular wave), produced by a pulse generator or by mixing sine waves, and containing only the odd-numbered harmonics (like the triangular wave, but differing in the relative strength of the harmonics)



envelope

When an instrument plays a note, the sound (the product of a specific waveform) does not simply begin and end abruptly. It has a shape, called its envelope, which consists of three parts:

- a beginning or attack—the time it takes for the sound to reach a given pitch and loudness
- a middle or sustain—a relatively steady state when the sound remains at a certain level of pitch and loudness
- an end or decay—the time it takes for the sound to die away

The envelope of a sound contributes to its distinctive quality or timbre. For example, a xylophone sound has a sharp attack, no real steady state, and an immediate decay:

When a gong is struck, the sound builds to its highest point, then gradually decays:

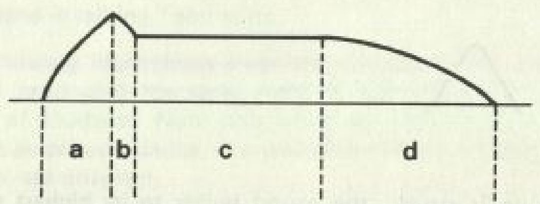
electronically controlled envelopes

The envelope of any natural or instrumental sound source is part of its characteristic sound. To create electronic music, however, the composer must design and control the envelope of every note in his composition.

Working directly with tape modifications, he can alter the sound's envelope by cutting and splicing to remove any part of the sound, or to reverse or rearrange the natural attack/sustain/decay sequence.

The synthesizer offers a far greater and more precise range of control over the envelope—an important area of sound modification that is discussed in the next section. Generally, how-

ever, it provides electronic controls over each individual phase of sound, dividing the envelope into four parts:



a) attack (as before)

b) decay—in this case defining a small, momentary (sometimes inaudible) fall-off from the attack peak

c) sustain (as before)

d) release—a re-naming and refinement of what has been traditionally called the decay: the time it takes for the sound to die away

(Synthesizer controls for this four-part sequence are usually labeled 'ADSR'.)

Since the envelope shape contributes directly to timbre, any modification—either through tape editing or electronic manipulation—will affect the waveform shape and alter the characteristic color of the sound it produces.

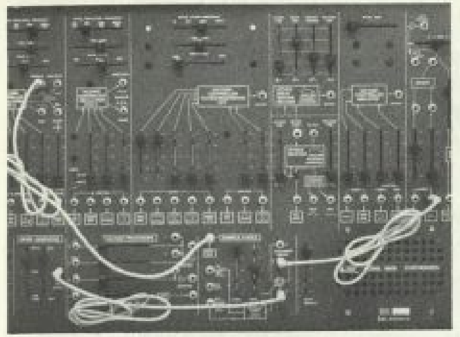
from acoustic to electronic vibrations

The plucked violin string is a typical example of a vibrating body that initiates vibrations (or oscillations) that are conveyed through the air to the ear.

Many sounds, however, are not conveyed directly from the original source to the ear. For example, when we listen to a radio broadcast, or music from records or tapes, we are really hearing sounds which have been reproduced by a different kind of vibration: the oscillation of an electrical circuit.

Electronic oscillations are not audible in themselves, but can be made audible by feeding them through a loudspeaker. In this process, the oscillations are converted into mechanical vibrations (the back-and-forth movement of the loudspeaker's cone) which, in turn, are conveyed through the air to the ear.

Electronic music is based on this very principle: that electronic oscillations can be used to produce sound. This is the province of the component called the oscillator or wave generator, and of the instrument that uses these components as the source of its raw sound materials: the synthesizer.



the synthesizer

what is it?

A synthesizer is a sound factory.

Like a macaroni factory that produces the raw material of pasta and processes it into endless and multiform varieties of macaroni, the synthesizer produces basic sound waveforms which are then added together, subtracted, cut, bent, and shaped into infinite varieties of finished sound.

In the same way that a macaroni factory is not a single invention, but rather is an aggregate of individual inventions (mixers, extruders, choppers, etc.), a synthesizer is not a single invention either. It is an aggregate of oscillators, filters, ring modulators, etc., which are devices that were developed individually over a period of time.

Recently, when enough components had accumulated to control all the elements involved in the creation of sound, they were consolidated into a single unit and called a synthesizer.

Even though the conglomerate now has a name, the individual components are still called oscillators, generators, filters, and so on. They fall into three categories:

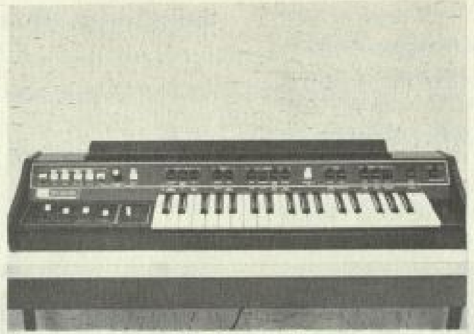
- sources, which produce the raw materials of sound in the form of basic waveforms;
- modifiers, which modify or alter these waveforms in a variety of ways; and
- controllers, which control or regulate the action of both the sources and the modifiers.

voltage control

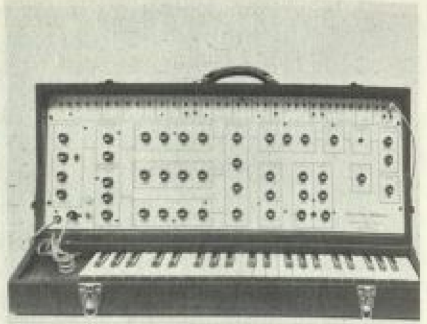
In the same way that pressure drives a flow of water through a pipe or garden hose, an electromotive force called voltage drives electric current through the circuitry of any appliance or instrument that uses electricity as its power source.

Working with the synthesizer, the composer can regulate a component by hand, or can take advantage of control voltage to do the same task with a degree of precision and speed impossible in manual control. For example, you can open and close a switch by hand, or turn a volume-control knob, but

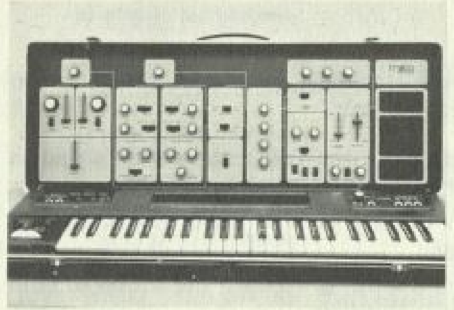
Moog Satellite



ARP Pro Soloist

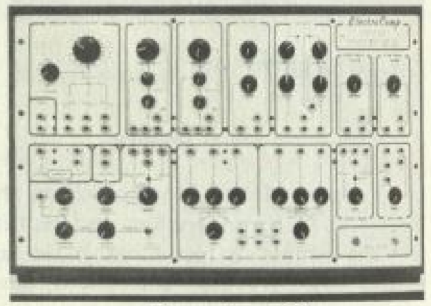


ElectroComp 101

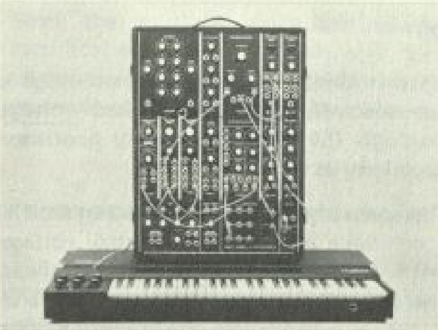


Moog Sonic Six

ARP Odyssey



ElectroComp 200



Moog 15

ARP 2600

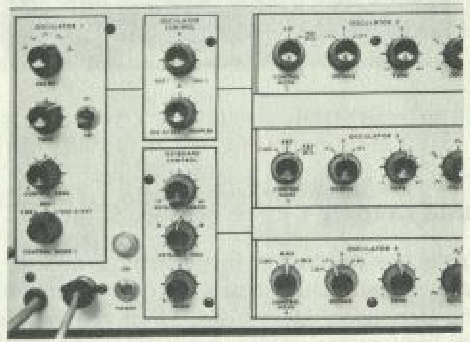
RMI Keyboard Computer

voltage can perform these operations at the rate of several hundred—or even several thousand—times per second. Since this driving force can be regulated, and can be applied in a vast number of ways to affect all the elements of sound—frequency, timbre, amplitude, and duration—the advantages to the composer are enormous.

Some of this choice (manual control versus voltage control) is determined by the nature of the particular component being used, since not all components produce voltage and not all components can be voltage-controlled. You must use a control-voltage source to regulate a voltage-controlled unit.

We will discuss this important interaction in greater detail. To begin, however, note the labeling on the synthesizer's panel for three of the most essential voltage-controlled modules: VCO, for the voltage-controlled oscillator; VCF, for the voltage-controlled filter, and VCA, for the voltage-controlled amplifier.

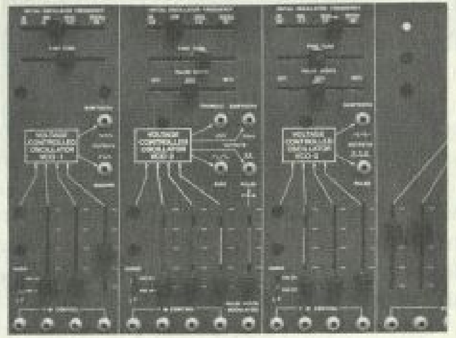
sound sources and processes



oscillators

The three main bodies of components used to synthesize sound are installed behind the control panel from left to right—the same path taken by the audio signal as it moves across the sound production line. In order, these are the sound sources, the sound modifiers, and the controllers.

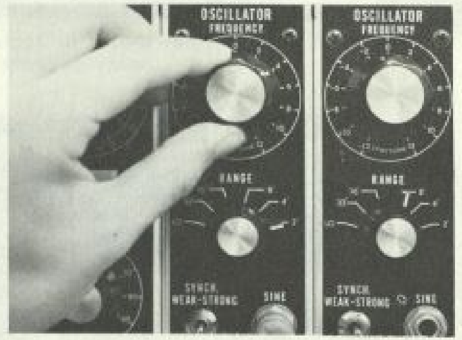
The synthesizer's principal sound sources (some of which may also function as voltage-control sources) are the oscillators. These produce four basic waveforms, with individual timbres, controllable frequencies, and fixed amplitudes.



The timbre is selected by the composer through his choice of waveform, either sine, triangular, sawtooth, or rectangular (see pp. 42-43).

In computerized synthesizers basic or complex waveforms are selected by inserting a punch card.

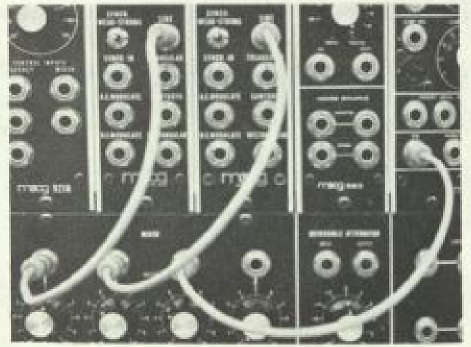
The frequency of the selected waveform (which can be regulated by manual or voltage control) can be anywhere in or out of the audible range: from sub-audio frequencies (below 20 Hz) to ultrasonic frequencies (above 16,000 Hz).



The amplitude of the selected waveform, a fixed output at the sound source, can be regulated only at a later stage of sound processing.

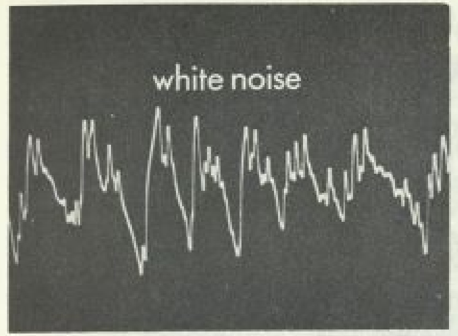
additive synthesis

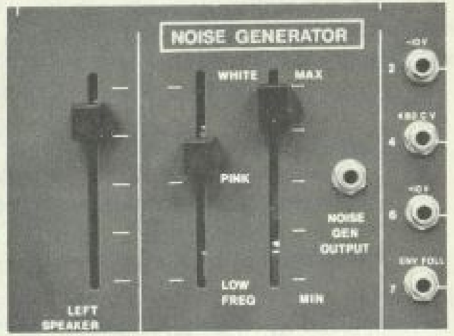
Although the composer can use these basic waveforms in their unaltered state, they are usually changed in some way. One way is to combine two or more waveforms using the synthesizer's mixer—a process called additive synthesis—to create a change in timbre. This produces new waveforms that can be custom-designed in various degrees of complexity. Generally speaking, the more complex and irregular the waveform, the sharper the sound.



noise generators

The noise generator produces a mixture of all audible frequencies. This mixture, called white noise, resembles the hissing found when tuning between FM stations. Since the ear is more sensitive to high frequencies, white noise seems high pitched. Some of the high frequencies can be filtered out to obtain a more audibly balanced mixture called pink noise. Noise is used mainly for percussive effects and to color other sounds.

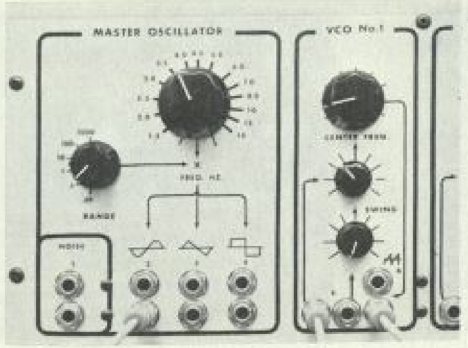




frequency modulation

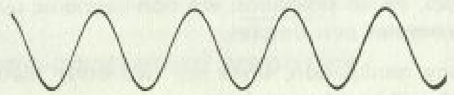
Modulation, in general, means control. In the case of sound sources, modulation means controlling the frequency, either manually or by control voltage from another source.

For example, the frequency of one oscillator signal can be voltage-controlled by the signal of a second oscillator. The one being controlled is called the audio signal, or carrier; the signal which does the controlling is called the modulator, or program signal.



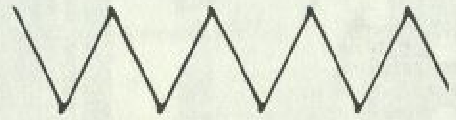
In this interaction of two signals, it is the modulator's waveform that determines the frequency change of the carrier. In fact, you can actually hear the controlling waveform if its frequency is sub-audio:

- A sub-audio sine wave



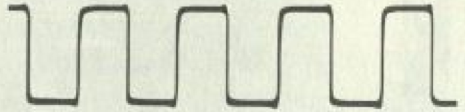
produces a frequency change in the carrier that smoothly rises and falls—in effect, a siren sound.

- A sub-audio triangular wave



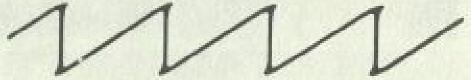
produces a frequency change that gradually rises, then immediately begins a gradual fall-off.

• A sub-audio rectangular or square wave



produces a frequency change that abruptly alternates between two frequencies, with no perceptible rise and fall.

• A sub-audio sawtooth wave

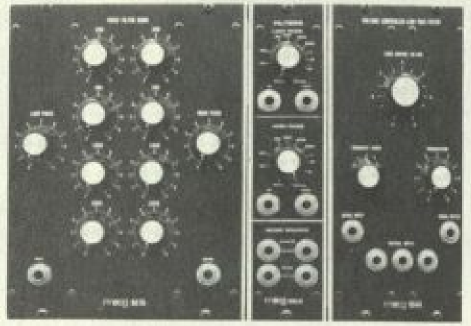


produces a frequency change that gradually rises, then abruptly falls.

If, however, you increase the frequency of the modulator into the audio range, you no longer hear its waveform but now hear an entirely different result: the generation of frequencies in addition to the carrier and modulator frequencies. These additional frequencies, called sidebands, are non-harmonic overtones which produce complex new timbres.

Besides frequency modulation, there are two other modulation techniques which will be taken up later.

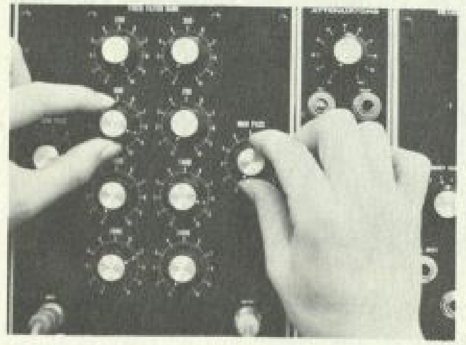
modifiers and processes

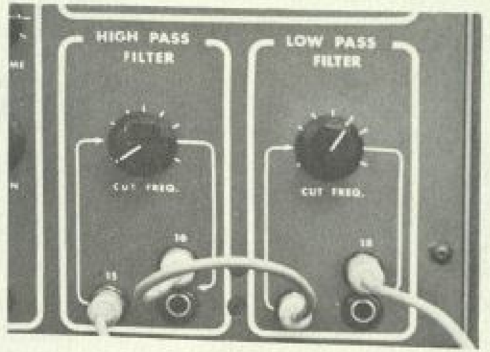


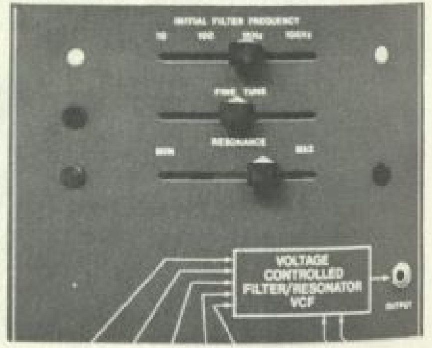
filters and subtractive synthesis

Sound sources can be used alone or in combination, as is or modified in a number of ways. One modification process, called subtractive synthesis, involves use of the synthesizer's filters.

Filters are modifiers that change timbre. They do this by subtracting any part of the frequency range of any sound above or below a variable cutoff point. Depending on the filter, the cutoff point can be manually controlled, or both manually and voltage controlled.







• The high-pass filter subtracts the frequencies below the cutoff point and passes those above.

• The low-pass filter subtracts the frequencies above the cutoff point and passes those below.

• The band-pass filter, with two cutoff points, subtracts both the high and low frequencies and passes the center frequencies.

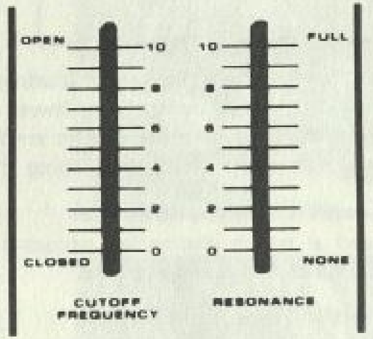
• The band-reject filter, also with two cutoff points, subtracts or rejects the center frequencies and passes both the highs and lows.

• The fixed filter bank divides the entire audio frequency range into a number of bands which can be subtracted from the whole, either individually or in any combination. It has manual controls to pass or suppress each individual band width.

Since filtration deals with a variable cutoff point, the composer is working with an adjustable element that can be regulated either by hand or by control voltage. We've already mentioned the superiority of voltage control in regulating components; now you can observe its advantages of speed and precision in controlling a specific module of the synthesizer. This is the voltage-controlled low-pass filter—the filter most frequently used in electronic composition because of its usefulness in passing the fundamental and altering the overtones.

If you were to vary the cutoff point by hand, moving it quickly up and down, you would be alternately passing and rejecting a certain band of overtones. This action alternately enriches and dulls the sound without changing the pitch, creating the familiar wah-wah sound.

However, you can get the same effect by using a sub-audio waveform from an oscillator as a control voltage. (Using different forms of control voltage—that is, from sources other than an oscillator—would produce different effects.) Besides being a more precise way to control the opening and closing of the filter, voltage control has the obvious advantage of freeing the composer's hands to manipulate other controls.

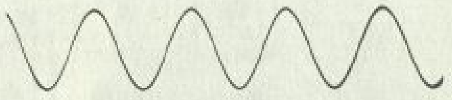


Another feature of the voltage-controlled low-pass filter is the resonance or regeneration control. This control emphasizes the overtones closest to the cutoff point, brightening the sound.

voltage-controlled amplifier

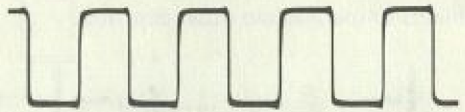
Using a voltage-controlled amplifier (VCA), the degree of loudness can be controlled either manually or by control voltage from various sources—an oscillator, for example. In the same way that you can hear the frequency change correspond to a sub-audio waveform, you can now hear the amplitude correspond to a sub-audio waveform:

• A sub-audio sine wave



gradually increases, then gradually decreases the loudness.

• **A sub-audio rectangular or square wave**

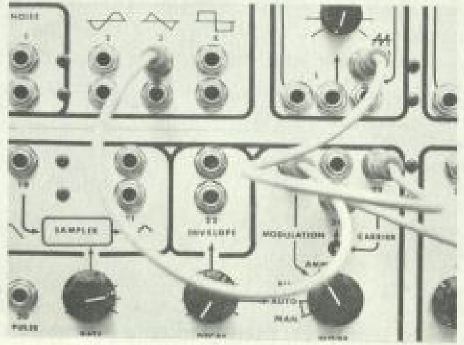


alternates between two degrees of loudness. Used in conjunction with manual control, the lower amplitude level can be set, if desired, to produce zero amplitude (silence). This would have the effect of turning the amplifier on and off.

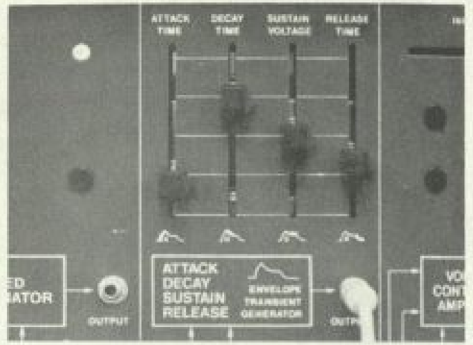
The rate of speed at which alternation occurs is determined by the frequency of the controlling waveform.

amplitude modulation

When you increase the frequency of the modulator into the audio range, you will hear additional frequencies. This modulation technique, called amplitude modulation (AM), produces fewer additional frequencies—and, consequently, a less complex timbre—than the similar technique of frequency modulation (FM).



controllers



envelope generator

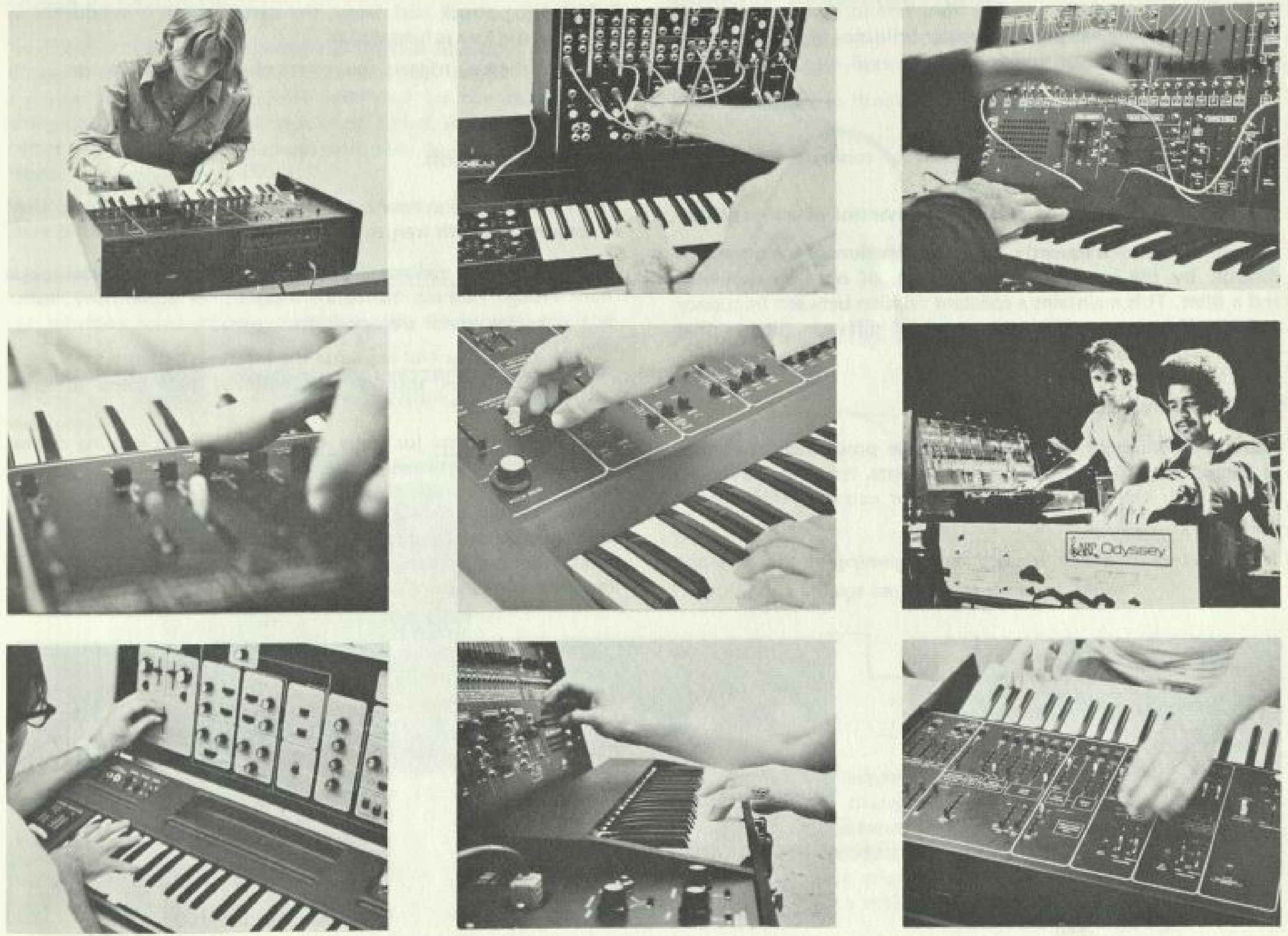
The most frequently used control-voltage source for a VCA—and probably the most important to the composer—is the envelope generator. The VCA and the envelope generator, working together, shape the sound's envelope by using the generator's separate controls for attack/decay/sustain/release (ADSR) (see pp. 43-44).

These controls, which determine duration for each envelope phase, can be set by the composer to produce a phenomenal range of envelope shapes: the envelope of any known sound, custom-designed shapes, even backward sound.

As a control-voltage source, the envelope generator can be used to control not only amplitude but frequency (by controlling an oscillator) and timbre (by controlling the filters).

keyboard

A synthesizer keyboard is different from a piano or organ keyboard. As a control-voltage source, each key when depressed releases a different pre-set voltage. As in the case of the envelope generator, this voltage can be used to control frequency, timbre, or amplitude, alone or in any combination.



Depending on patching, moving from left to right on the keyboard produces higher pitches, greater brilliance of a single pitch, or a progressively louder sound:

- The increase in pitch level is a result of frequency control of an oscillator.
- Increased brilliance is a result of raising a filter's cutoff point.
- The louder sound results from control of an amplifier.

One of the most frequently used combinations is simultaneous control, by the same keyboard voltage, of both an oscillator and a filter. This maintains a constant relation between frequency and cutoff point, producing a range of different pitches that have the same timbre.

using the keyboard as a trigger

Another function of the keyboard is the production of voltage to trigger the action of certain components. When the keyboard is used with the envelope generator, for example, this is what happens:

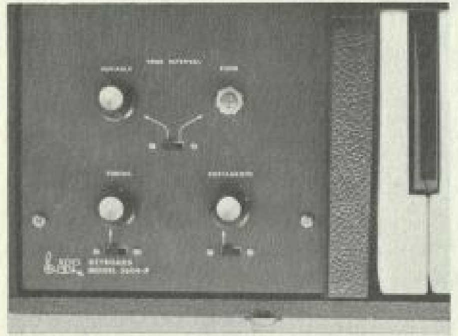
1. Depression of a key triggers the beginning of the envelope cycle.
2. Following attack and decay, the sustain state is maintained as long as the key is held down.
3. Lifting the key triggers the release phase of the envelope.

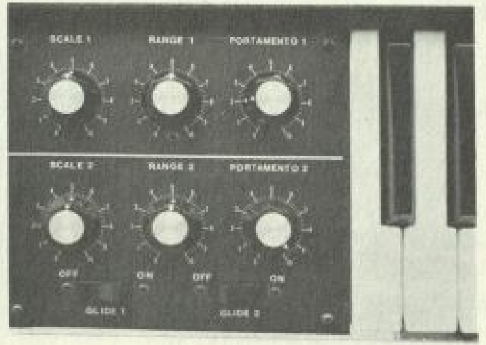
keyboard controls

The synthesizer keyboard commonly has three controls. Used in conjunction with frequency, they function as follows:

- A tuning control regulates pitch over an extremely wide range: tuning to concert pitch, to accompany instruments, or for transposition.
- A scale control regulates the intervals between keys: from microtones through conventional half steps to larger intervals.
- Portamento (or glide control) regulates varying degrees of glide between pitches.

These controls produce a similar effect on timbre and amplitude. Microtone tuning, for instance, can be used to produce minute timbre variations; or the portamento, used with an amplifier, can produce crescendos and diminuendos.





ribbon controller

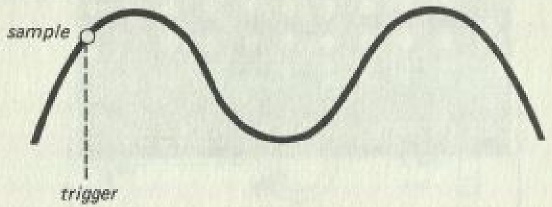
The ribbon controller is a metal ribbon that produces a varying voltage when you move your finger along its length. The result is similar to the gliding effect produced by the portamento setting of a keyboard, but the more direct means of control enables you to produce minute variations as well as sweeping effects.

foot pedal

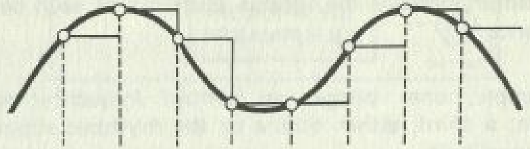
A foot pedal produces a control voltage which is varied by foot pressure. Although the degree of control is less precise than that provided by a ribbon controller, the foot pedal has the advantage of freeing the composer's hands for other functions. This is of particular advantage in live performance where subtlety of control is frequently sacrificed to speed and flexibility.

sampler, or sample/hold

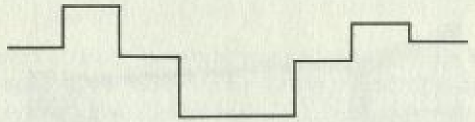
A sampler produces a sequence of control voltages. When it receives a trigger impulse it samples the voltage level of a waveform:



and holds this voltage level until it is triggered to make the next sample:

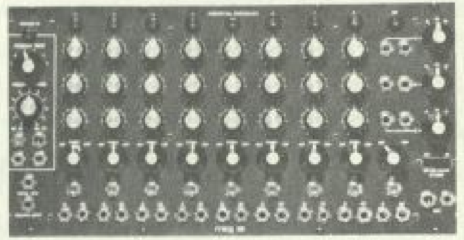


The result is a voltage sequence of differing levels:



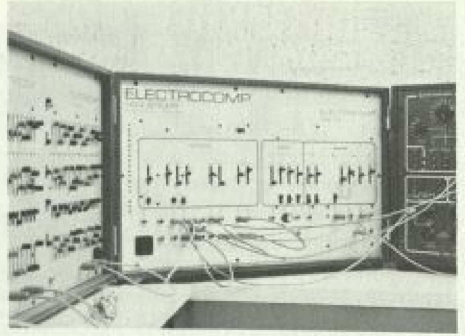
The form of the sequence is determined by the form of the sampled wave. A triangular wave will produce a sequence of increasing and decreasing levels which, if used to control an oscillator, will produce ascending and descending pitch sequences resembling musical scales. A random waveform, like noise, will produce a sequence of random voltage levels.

sequencer

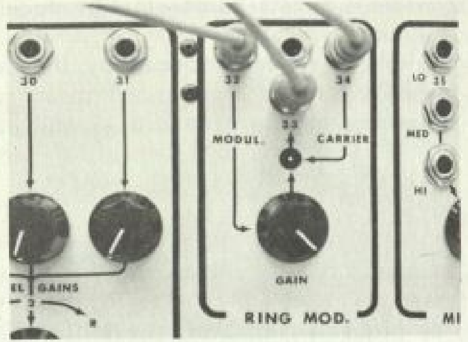


The sequencer is a complex controller which gives the composer precise control over a number of aspects of a sequence of pitches. This is possible because each stage of the sequence produces up to three separate voltages which can be pre-set to control simultaneously the various elements of each sound in the sequence.

For example, one voltage can control frequency; another, amplitude; a third, either timbre or the rhythmic structure of the sequence itself.

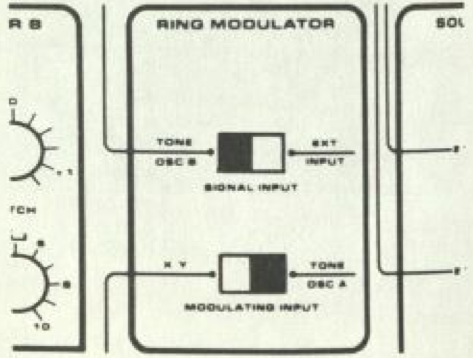


miscellaneous components



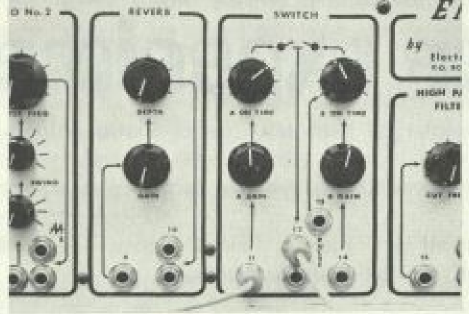
ring modulator

Ring modulation produces an effect closely related to amplitude modulation. Whereas AM produces frequencies in addition to the frequency of the signal being modulated, ring modulation produces the same frequencies but eliminates the original signal. The result is an unusual, rather bizarre timbre.



reverb

The synthesizer's reverb unit artificially creates any degree of reverberation. Like any other modification technique, reverb can be applied to signals either produced by the synthesizer or from external sources.



electronic switch

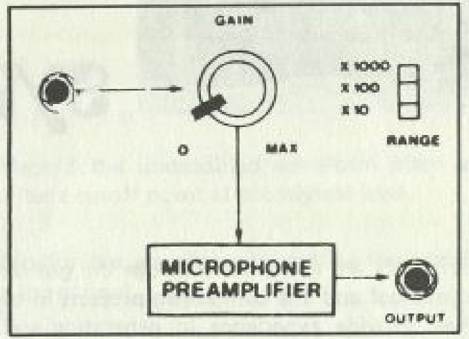
The switch interrupts the flow of a signal by opening and closing its circuit.

Switching a single signal has the effect of chopping holes in the sound by alternating sound and silence, breaking continuity or flow. How effectively this alters or masks the sound's identity depends on switching speed.

Switching two signals creates an alternating sequence—a kind of revolving-door effect—which, if done rapidly enough, modifies the color of both signals by producing a third, composite, timbre.

microphone pre-amplifier

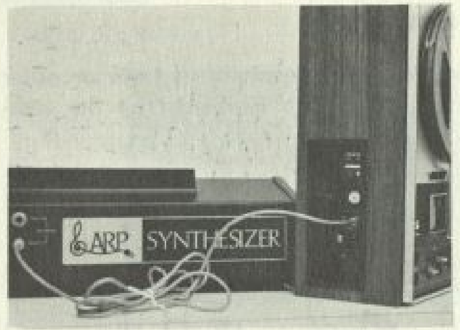
The mic pre-amp brings low-level signals—from mics or other low-level external sources—up to the level used by the components of the synthesizer.



High-level signals—from tape recorders, for instance—may be routed directly to any component without being pre-amplified.

An unlimited array of external sources are available to the composer for synthesizer modification. Any sound that can be taped or picked up by a mic can be modified. Sounds from radio, TV, a record player, as well as any electronic instrument (electric guitar, electric piano or organ, another synthesizer, etc.) can be fed directly to the synthesizer through a line-input, then modified.

These modifications include filtration, amplitude modulation, ring modulation, switching, reverb, as well as the imposition of envelope shapes on the external signal.



synthesizer experiments

These experiments are designed to bridge the gap between your synthesizer manual and the composing projects in the following section. They provide experience in generating and combining sounds within the structure of a basic compositional form. The form consists of both sequential and simultaneous events. Emphasis is largely technical in the first experiments. The later ones provide increasing opportunities for creativity.

equipment

Synthesizer with keyboard, and stereo tape recorder. A separate mono or stereo playback machine is required for some experiments and is optional for others. A quadraphonic recorder is optional for the last experiment.

format

Most of the experiments consist of a series of short paired events which are taped in this order: Events 1, 2, and 3 are recorded on Track 1; the tape is rewound, and Events 4, 5, and 6 are recorded on Track 2. Paired events, 1 with 4, 2 with 5, and 3 with 6, will be heard on playback.

procedure

1. Familiarize yourself with the purpose and patching of each experiment and with the character of each event.
2. Patch the synthesizer components as directed.
3. Experiment with programming the synthesizer to produce the desired sounds.
4. Record Event 1 for ten to thirty seconds. A similar duration will be adequate for each event. In most cases the tape must be stopped between events in order to re-program the synthesizer. In cases where programming is unchanged, proceed directly to the next event.
5. Record the remaining events on Track 1. Rewind.
6. Monitor Track 1 while programming and recording Track 2.

The approximate synchronization of events produced by this method of recording (sound-with-sound) is adequate for these experiments. Optionally, sound-on-sound may be used to mix the two tracks in exact sync onto one track, or separate tracking with exact sync can be achieved by playing back Track 1 on a playback machine and re-recording it while recording the new events on Track 2. (See Combining Sounds, pp. 28-29.)

Since these experiments are primarily for exploration, and only secondarily for composition, editing is optional. The finished results, however, should be saved for later use as possible composition elements.

If your synthesizer has only internal connections (which predetermine patching possibilities), you may be unable to patch it as indicated in certain experiments. If so, try to adapt the experiment to the capabilities of your instrument by substituting components or circuitry.

experiment 1

purpose

to observe how the timbre of a waveform is altered by varying the filter's cutoff point

patching

oscillator → low-pass filter → (to amplifier, or through mixer to amplifier)

sound source

- sawtooth or rectangular wave

modifications

- frequency: one fixed frequency for Track 1, another for Track 2 (Don't play the keyboard.)
- timbre: altered by manually varying the cutoff point of a low-pass filter
- amplitude: fixed throughout

track 1

Event 1: Record the unmodified waveform after setting the filter's cutoff point at the highest level.

Event 2: Modify the waveform by setting the cutoff point at a lower level.

Event 3: Modify the waveform by manually varying the cutoff point while recording.

track 2

Event 4: Before recording, and while monitoring Event 1, experiment to find another frequency of the same waveform—and a new setting of the cutoff point to accompany Event 1.

Event 5: Keep the new frequency, but choose a different cutoff point to accompany Event 2.

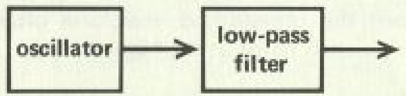
Event 6: Choose one or more new cutoff points to accompany Event 3.

experiment 2

purpose

to observe how the timbre of a waveform is altered by varying its frequency in relation to a fixed position of the filter's cutoff point

patching



sound source

- sawtooth or rectangular wave

modifications

- frequency: manually varied by using the oscillator's frequency control knob (Don't play the keyboard.)
- timbre: altered by frequency changes in relation to a fixed position of the low-pass filter's cutoff point
- amplitude: fixed throughout

track 1

Event 1: One waveform set at a low frequency. Set the filter's cutoff point at mid range.

Event 2: Keeping the same cutoff point, alternately raise and lower the frequency. End the event on a frequency that is higher than the one you began with.

Event 3: Continue the same higher frequency, but reset the cutoff point at a higher level.

track 2

Event 4: Set the cutoff point at a middle level and, using the same waveform, alternately raise and lower the frequency. End the event on a low frequency.

Event 5: Set the cutoff point at a new position, and continue the same low frequency.

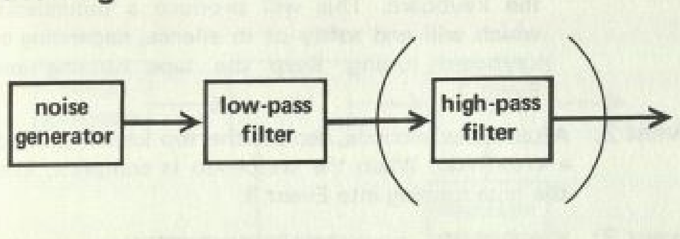
Event 6: Keeping the same cutoff point, alternately raise and lower the frequency.

experiment 3

purpose

to observe how filtration alters the timbre and pitch of noise

patching



sound source

- noise generator

modifications

- timbre and pitch: altered by varying the cutoff point of a low-pass filter (optionally combined with a high-pass filter)
- amplitude: Track 1, fixed; Track 2, variable through use of the tape recorder's level control

track 1

Event 1: Filter the noise, using the low-pass filter with a low cutoff point.

Event 2: Alternately raise and lower the cutoff point. End the event with a high setting.

Event 3: Keeping the same high cutoff point, gradually fade out the sound with the tape recorder's level control.

track 2

Event 4: With a low recording level, rapidly vary the cutoff point. End the event at a low cutoff point.

Event 5: Keep the same low cutoff point, but vary the recording level from soft to loud in an irregular time pattern.

Event 6: Starting softly, at the same low cutoff point, gradually raise the recording level and simultaneously begin to vary the cutoff point.

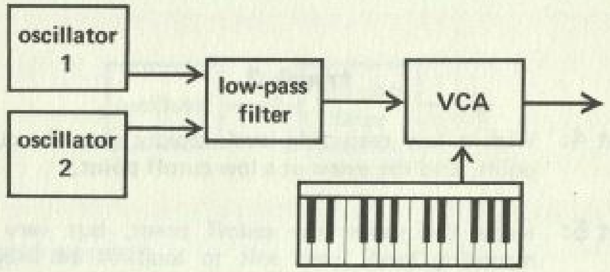
Optional: If you have both a high-pass and low-pass filter, combine them as a band-pass filter for Events 4, 5, and 6. You can try to keep the band width constant, or you can vary the width at any time. If you are working alone, you will have to omit the volume changes since three controls are being manipulated at the same time.

experiment 4

purpose

to observe how the voltage-controlled amplifier (VCA) can be controlled by the keyboard portamento

patching



sound sources

- oscillator 1: sawtooth wave
- oscillator 2: rectangular wave

modifications

- frequency: manually variable using the oscillators' frequency controls
- timbre: constant, using a fixed high setting of the filter's cutoff point
- amplitude: varied from high to low by alternately depressing the keyboard's top and bottom keys

track 1

Event 1:
1. Depress the top key of the keyboard to initiate maximum amplitude. Set the portamento control at maximum.
2. Tune both oscillators to the center frequency range. They may be tuned to the same frequency or to a close interval. Start recording.
3. After a few seconds, depress the bottom key of the keyboard. This will produce a diminuendo which will end softly or in silence, depending on keyboard tuning. Keep the tape running into Event 2.

Event 2: After a few seconds, depress the top key to produce a crescendo. When the crescendo is complete, keep the tape running into Event 3.

Event 3: Keeping the same amplitude level, vary the frequency settings of the two tones by gradually raising one and at the same time lowering the other.

track 2

Event 4: With the oscillators' frequency settings unchanged (same as end of Event 3):
1. Depress a low key, then start recording.
2. Gradually bring both tones toward the same center-range frequency.
3. A few seconds after you begin this frequency change, quickly depress the top key then immediately continue the change.

Event 5: Keeping the same high amplitude level, change the frequencies by gradually and simultaneously raising one and lowering the other.

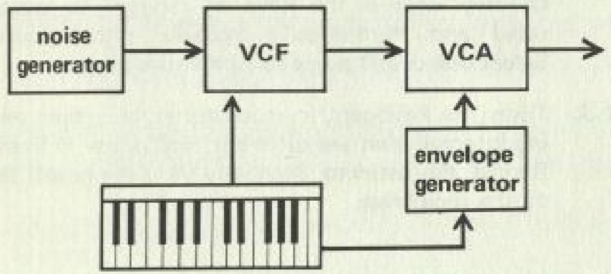
Event 6: Keeping the same high/low frequencies (end of Event 5), depress the bottom key after a few seconds.

experiment 5

purpose

to make a one-track tape loop (to be used in Experiment 8) consisting of bursts of pitched noise

patching



sound source

- noise generator

modifiers

- voltage-controlled filter (VCF)
- voltage-controlled amplifier (VCA)

controllers

- envelope generator
- keyboard

In this experiment, the keyboard controls the VCF in order to vary the pitch of the noise, and triggers the envelope generator to shape the sound.

procedure

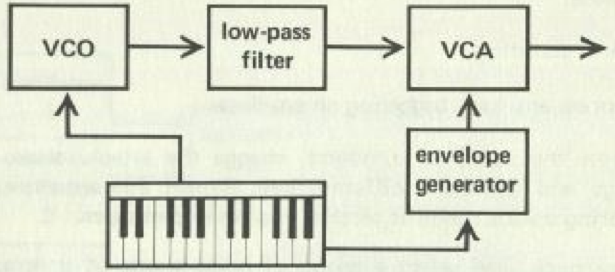
1. Set up patching.
2. Set filter cutoff point at mid range; set resonance in upper range.
3. Set envelope generator for short attack and moderately long release.
4. Start recording.
5. Depress any key, triggering an envelope.
6. When the decay is complete, change the attack/release settings and depress a different key. Repeat this sequence, exploring a wide range of pitches and envelope shapes.
7. Play back, and select a group of noise events of contrasting pitch and envelope.
8. Edit the tape to arrange the selected events into a series. (You can delete unwanted events, shorten events, rearrange portions of events, or add silence at any point.)
9. Make a loop of the edited tape, then play it back at various speeds.

experiment 6

purpose

to create pitch groups of varying speed, tuning, and timbre, using the keyboard

patching



sound source

- one voltage-controlled oscillator (VCO), producing one basic waveform: either sawtooth or square wave

modifiers

- low-pass filter
- voltage-controlled amplifier (VCA)

controls

- manual control, in some events, of the filter's cutoff point
- keyboard

In this experiment, the keyboard (through control of the VCO) regulates the frequency of the various pitch groups. Keyboard tuning is unconventional throughout, alternating between microtones (intervals smaller than the half step) and macrotones (intervals larger than the whole step).

track 1

Event 1: Tune the keyboard in macrotones. (If your keyboard doesn't have macrotone tuning, tune in microtones and play large intervals.) Create a repeated rhythmic figure (ostinato) using any two tones. Manually vary the filter's cutoff point during the ostinato.

Event 2: Tune the keyboard in microtones. Quickly depress the keys at random to create a rapid and rhythmically irregular pitch sequence. Select one cutoff point for the entire phrase.

Event 3: Tune the keyboard in macrotones, this time selecting intervals that are different than those in Event 1. Repeat the ostinato. Manually vary the cutoff point during the phrase.

track 2

Event 4: Microtones. Create a phrase of seven or nine different pitches, played slowly. Select one cutoff point for the entire phrase.

Event 5: Macrotones. Create a phrase of seven or nine different pitches by depressing the same keys used in Event 4. (The change in tuning will change the pitch.) Either select one cutoff point for the entire phrase, or manually vary it as you play.

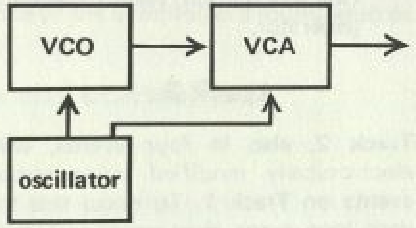
Event 6: Microtones. Approximate the tuning, rhythmic effect, and cutoff point of Event 2.

experiment 7

purpose

to create an oscillating texture exploring the various tremolo/siren effects of a sub-audio sine wave modulating an audio signal

patching



sound source

- VCO sawtooth wave (audio signal)

modifier

- VCA

controllers

- oscillator: sub-audio sine wave (modulating signal)
- manual regulation of frequency controls and attenuators for both oscillators

In this experiment, the sub-audio sine wave is used to modulate both the VCO (frequency modulation) and the VCA (amplitude modulation). This combination produces effects ranging from a slight tremolo to an exaggerated siren sound.

track 1

Event 1:
1. Set the VCO's sawtooth wave to a mid-range frequency. Set the oscillator in the sub-audio range, about two or three cycles per second (2-3 Hz) with the attenuator—the control that regulates the amount of signal being fed in—in the 'off' position.
2. Begin recording the audio signal. After a few seconds, use the attenuator to fade in gradually the modulating signal from 'off' to maximum. This will produce a progressively increasing siren effect.
3. After a few seconds, gradually fade down to a moderate degree of modulation.

Since you are using the same sub-audio waveform to modulate both the VCO and the VCA, the frequency and the amplitude will increase and decrease together.

Event 2: Gradually increase the frequency of the modulating signal until it approaches, but does not enter, the audio range. This change will slowly increase the rapidity of the siren oscillations.

Event 3: Use the attenuator to increase the degree of modulation to maximum. After a few seconds, simultaneously decrease both the degree and the frequency of the modulating signal.

track 2

Event 4:
1. Reset the frequencies of both the audio and the modulating signals. For the audio signal, choose a frequency that is different from that used in Event 1. Set the modulating signal at a frequency similar to that of Event 1. At the start of Event 4, the attenuators for both oscillators will be in the 'off' position.

2. Monitor Event 1 to locate the beginning of the frequency modulation fade-in. At this point, begin recording a gradual fade-in of the new audio signal.
3. After a few seconds, begin a slight amplitude modulation by fading the modulating signal into the VCA. This will create a slight tremolo.

Event 5: Continue the tremolo. Midway through this event, gradually fade out the modulating signal until the audio signal returns to a fixed amplitude.

Event 6: Continue the fixed-amplitude signal. After a few seconds, simultaneously fade in the amplitude modulation while increasing its frequency to maximum. (Depending on your instrument, 'maximum' will either be at the threshold of the audio range or into it.)

experiment 8

purpose

to create textural material combining noise and electronically modified vocal sounds

sound sources

- the pitched-noise tape loop made in Experiment 5
- modified voice sounds (using the synthesizer's mic input)

modifiers and controllers

- optional

track 1

Events 1-4: For Track 1, create a series of four events using the tape loop as your sound source. For each event, modify the original sound in a distinctive way, using the capabilities of both the synthesizer and the tape recorder in any combination:

- recorder—speed change, backward loop, echo, etc.;
- synthesizer—filtration, amplitude, modulation, ring modulation, reverb, switch, and envelope generator.

track 2

Events 5-8: Track 2, also in four events, will consist of electronically modified voice responses to the events on Track 1. To make this track, monitor each loop event, then create a vocal "answer" that in some way reflects or reacts to the bursts of noise. Use any appropriate synthesizer modification of a wide range of voice sounds:

- speech (sentences, vowel sounds, nonsense syllables);
- song (complete or fragmented tunes);
- imitations of natural or electronic sounds;
- cries, whispers, shouts, laughter, sobs, etc.

Exact synchronization is preferable for this experiment—use sound-on-sound or dub Track 1 while recording Track 2.

The microphone hook-up can be made through the synthesizer's mic-amp input, or by using some sort of mic pre-amplifier—another tape recorder, for example.

experiment 9

purpose

to create a three-layer texture, combining sounds from both electronic and non-electronic sources

sound sources

- a tape of natural, instrumental, or machine sounds
- any one of the synthesizer's sound sources

modifiers and controllers

- optional

tracks 1 & 2

This experiment consists of three layers of sound on two tracks:

- natural, instrumental, or machine sounds;
- an electronic modification of these sounds;
- an electronic response to the combined events of the first two layers.

In choosing material for the first layer, keep in mind that this sound has to interact with two subsequent layers of material.

- It does not have to be especially interesting in itself. Simplicity in the first layer will balance the complexities of the second and third.
- Its sound should be thin. A thick or rich texture will interfere with the clarity of the accompanying material.
- It should be loosely knit, with a number of silences interrupting the sound. A hole in the first layer not only leaves room for a later event, but momentarily thins out and gives variety to the total texture of the finished tape.

The second layer of sound is created by modifying the original with the synthesizer. The general procedure and the alteration possibilities are the same ones used for the tape loop in Experiment 8: filtration, amplitude modulation, ring modulation, reverb, use of the switch and the envelope generator.

The third layer, made of electronic sounds only, should grow in the same way as the modified voice track in Experiment 8: as an answer that somehow reflects or reacts to the combined events of the first and second layers.

recording procedure

Layers 1 and 2

1. Place the original tape on the playback machine patched through the synthesizer to Channel A of the recorder.
2. Before recording, experiment with a variety of electronic modifications of the original tape. When you have what you want, record it. (You can do this piecemeal or in one uninterrupted session.)

Combining Layers 1 and 2

3. Play back the original and the modified version at the same time. Sync the opening sounds, then listen carefully for

possible changes that will improve the total sound. This might include deletions, rearrangement of material, addition of silence, a different alignment of the two tracks (starting one tape before the other), and so on. If you've misjudged the effect of certain modifications on the combined sounds, tape one or more alternatives and splice them in, or substitute a completely new version.

4. Combine both layers on Channel B (sound-on-sound).

Layer 3

5. Play back the combined tracks on the playback machine to determine what kind of electronic sounds would combine well with them. Then experiment with various possibilities (without recording) while the tape is running. When you have an approximate idea of what you want, record it on a separate tape; this will allow you to make alternate versions and to edit the electronic sounds to fit the other layers.

Adding Layer 3 to the stereo master

6. When the new layer is in final form, either dub it on the open track of the stereo master in rough sync with the sound-on-sound track, or play back both tapes on separate playback machines to record them in exact sync on a new tape.

experiment 10

purpose

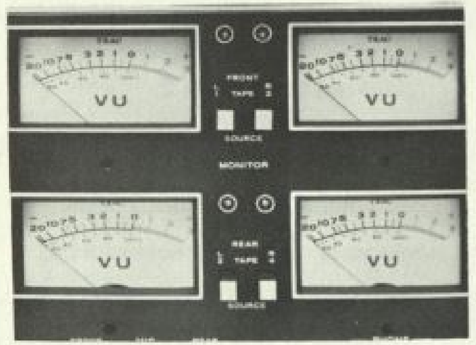
to explore the sound possibilities of multi-track textures, using electronic sources alone or in combination with vocal, instrumental, natural, or machine sounds

sound sources

- electronic, others

modifiers and controllers

- optional



tracks 1 to 4 (or more)

This experiment differs from Experiment 9 in the greater number of sound layers that are combined to make the electronic texture and, consequently, in the greater number of options for selecting the sounds themselves. Otherwise, the basic idea—building a rich texture by combining separate layers of sound—is the same.

If you can, use a quadraphonic recorder (although stereo with sound-on-sound is an adequate alternative). Starting with four tracks, you can add additional layers by mixing down: three tracks mixed down to one, for example, leaving the first three open for later additions. In general, work for clarity.

- Keep the material simple. Complexity occurs in the interaction of the multiple lines.
- Allow for intermittent silences or fade-outs in a given layer. This varies the thickness of the overall texture, and leaves room for later events to be heard.
- Choose distinctive timbres for each layer, since lines of similar color (especially in close proximity) tend to confuse the ear.
- Assign different sounds to different frequency bands. For example, a four-layer texture could contain these sounds:

  high register—random samplings of a waveform

  high-medium—modified voice sounds

  medium—a fixed pitch sequence

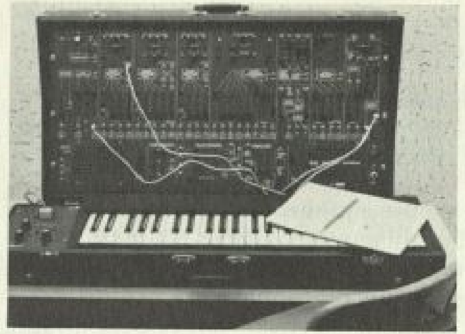
  low—a rhythmic pattern (white noise with a percussive envelope), or synthesized string bass line

  This sort of spatial separation keeps sonic identities intact, and allows the ear to follow one sound moving in one register.
- Follow the same principles used by orchestrators. Building an electronic texture is a form of orchestration.

Thick, rich, or loud low sounds need reverberation space. Keep other layers out of their frequency band.

Two or more layers of thin sounds—or comparatively soft ones in the lower frequency range—can move compatibly in the same register.

Keep events simple and spacious in the low register, with ample breathing room; reserve the upper registers for complex sounds, rich timbres, rapid figurations, and the mixture of multiple lines.



composing projects

how to start

Begin your composition with a basic idea. This might be a single sound you like, one of your experimental tapes, a group of pitches, an interesting rhythm, or a specific modification technique you'd like to explore. The idea for the composing project might stem from a non-musical source such as a light or slide show, a poetry reading, or a play, dance, or film that you plan to accompany with music.

Your next step is to think out a number of possibilities for developing, expanding, and exploiting the basic idea. For example:

- A taped event can be played forwards or backwards; altered by speed change; modified in timbre; edited to change its duration, sequence, or envelope; made into a loop; or played against itself in a multi-layered texture.
- A pitch group can be recorded forwards, backwards, or in varying pitch sequences; modified by speed, timbre, or register changes; made into layers of simultaneous but contrasting loops; and—if played on a synthesizer keyboard—the intervals can be varied from microtones to macrotones.
- A rhythmic idea can be stretched, condensed, fragmented, or looped; and recorded on different pitch levels with new colors and varying envelopes.
- A modification technique—such as reverb or filtration—can be applied to a succession of events produced by one sound source, or by several related or contrasting sources.
- Words or staged events are especially helpful. They suggest mood, sequence, durations, and the function (background, foreground) of your music.

how to continue

Try to keep a relaxed attitude toward the project, and an open mind about new developments and unexpected changes that turn up in the course of an experiment. Don't lock yourself into one way of thinking.

Don't concern yourself about the length of the composition. A thirty-second or one-minute piece can convey a lot.

If you can't get started, make a list of experiments you can run with one of your favorite tapes. Then try out new ideas that come out of the experiments, and new ways of editing and combining the sounds you have. Work by trial and error until you begin to develop interesting material.

If you get stuck, bored, or frustrated, or run out of ideas, try to be more objective about the project. Stay away from it for a while so that you can think things out and isolate your problems. If you're still dissatisfied, work out variations of the same tape, fresh combinations of material, unplanned modifications, or new sound sources.

If the basic material seems dull, expand the scope of your experiments. Flat, static, or unpromising sounds may yield surprisingly good results if you open up new paths of exploration.

If the piece itself is monotonous, redesign your sounds with more variety. You can create tension by using:

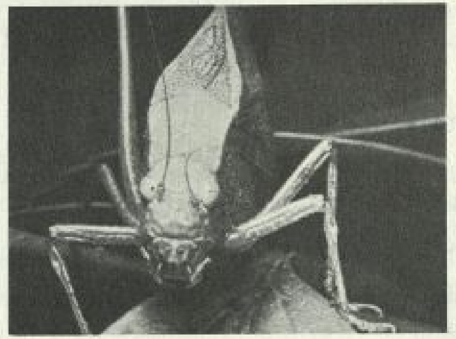
- fast tempos, accelerandos
- crescendos to high, loud pitches
- sudden silences
- thick or abruptly shifting textures
- unusual timbres
- extreme frequencies
- sharp contrasts of material

If the music is too compact, and seems to rush too quickly through the event sequences, you can create more relaxation by using:

- slow tempos, ritardandos
- diminuendos
- expected silences
- thin or gradually shifting textures
- conventional timbres
- soft sounds in the medium and low range
- repetition
- subtle changes of character from one event to the next

If the music seems shapeless, or tends to wander aimlessly, try to reorganize your materials into sections which, although related from one to the next, have a definite character and a clear shape. One section, for example, might contain only unusual timbres, rapid pitch sequences, or clangorous bell sounds. This, in turn, might be tied in with the next section (also with its own character) by threading an occasional appearance of one of those sounds through the new texture. If you work this way—or in any way that combines unity with variety—the music will hang together.

composition processes and materials



sound sources

natural or machine sounds

- bird song, insect and animal noises, plucked rubber

band, creaking door, clanking silverware, bottles (struck) and glassware (tapped or wet-rubbed on the rim), water sounds (in a pan, down the drain), comb (thumbnail down the teeth), rubbed balloon, crumpled cellophane, finger-snapping, finger-tapping (on the mic)

- auto horn, air brakes, motors, jack hammer, fan, vacuum cleaner, drill, electric toothbrush, typewriter (keys, sliding carriage, bell)

instrumental sounds

- winds: piccolo, flute, clarinet, bassoon, saxophone, recorder, kazoo, whistle, toy horn, harmonica
- brass: trumpet, horn, trombone, tuba
- strings: guitar (acoustic or electric), ukulele, banjo, autoharp, violin, viola, cello, bass, harp
- percussion: drums, cymbals, claves, maracas, steel drums, pots and pans, metal lids, dry seeds in a glass jar, gongs, bells, toy rattles, party noise-makers, slapsticks, hollow gourds
- keyboard instruments: piano, organ, harpsichord, marimba, xylophone, glockenspiel, vibraphone, toy piano

Any instrument can be played conventionally, or explored for unusual effects. For example, you can blow through a detached brass mouthpiece; click the keys of a woodwind or the valves of a brass; tap on the body of an instrument; experiment with homemade mutes; strum or pluck the strings inside a piano, or change its timbre by inserting bolts, rubber wedges, or other objects between the strings; shout into the resonating chamber of a piano with the damper pedal down; slide a smooth object up and down the high strings of an electric guitar; explore the electronic modifiers used by rock groups (wah-wah pedal, fuzz box, reverb).

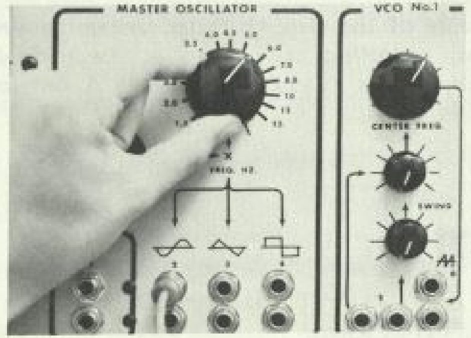
vocal sounds

- song: whole or fragmented tunes, humming, improvised melodies, vocal exercises
- speech: a prepared or improvised talk, selections from a poem or play, conversation, dubs from a radio or TV show, nonsense syllables

- other vocal and mouth sounds: tongue-clicking, lip-smacking, breathing, whispers, shouts, sobs, laughter, sighs, cries, gurgles, grunts, moans, imitations of natural or electronic sounds

electronic sounds

- synthesizer or separate components: oscillators, noise generators, random sampler; radio or TV static, white noise between FM stations, electrical system noises (hiss, crackle, hum), feedback



modifications

tape techniques

- speed change, backward sound, echo, feedback, panning, tape loop, sound-on-sound, sound-with-sound, hand manipulation, tape editing

electronic modifications

- frequency modulation, amplitude modulation, ring modulation, filtration, reverb, envelope imposition, electronic switching

source mixtures

Sounds from any sources can be combined either horizontally (in sequence in the same layer) or vertically (simultaneously in different layers). You can intermix different modifications of the same sound source, different sources from the same category (two or more natural sounds, for instance), and sounds from different categories (such as instrument with voice and electronic sounds). The compatibility of sounds, however, is something you will have to determine experimentally. What might seem to be a good combination may not work at all—or may work better than you thought possible.

sounds in formal structures

project 1

a piece in A-A^1 form

A-A^1 is a two-part form consisting of a section and its modified repetition. The two sections are usually of about the same length.

- A: A series of related events, long enough (at least one minute) to establish a statement with a definite character. The section should convey a feeling of beginning/middle/end: introduction of material, development, climax or relaxation.
- A^1: Section A with minor alterations—same general material with new modifications (timbre, frequency, etc.), duration changes (shorter, longer events; added, subtracted silences), or sequence changes (rearrangement of events). The easiest way to create A^1 is to dub, then edit, all of A.
- Bridge between A and A^1: Although your material may dictate a direct splice from A to A^1, the two-part form is clearer if the halfway point of the piece is marked in a definite way: silence, fade-out/fade-in, climax sound, unusual timbre, etc.

project 2

a piece in A-B form

A-B is a two part form consisting of one section followed by a second section of contrasting or opposite character. The two parts are usually of about the same length.

- Section B should have its own distinctive character. Create it by using modifications which either did not appear at all in A, or were only briefly suggested (one way of binding dissimilar parts of the same piece).
- Because of the clear character change between the two parts, a feeling of cadence at the midpoint—climax, repose, fades, silence—is optional: A can lead directly into B.

project 3

a piece in A-B-A form

A-B-A is a three-part form consisting of two sections of similar character separated by a middle section of contrasting character.

- To make this form work, be especially careful of sectional balance—a matter that has less to do with equal durations than with the nature of your material. For example, a powerful middle section may need longer development than the end sections; or a weakly developed opening (first A) may call for a richer, broader restatement at the end (second A); or a brief, wrap-up finale may be just enough to suggest and balance a strong, lengthy first section.

project 4

a piece in rondo form

The rondo is a form that deals with the frequent recurrence of a theme. (In electronic music, a theme could be a developed sound event with an identifiable character—one we would recognize when it returned.)

- If A represents the theme, B a contrasting interlude, and C yet another interlude with its own character, then a typical rondo form could be outlined as

  A-B-A-C-A

  or, even longer and with modified recurrences, as

  A-B-A^1-C-A^2-D-A

  or as

  A-B-A^1-C-B^1-D-A^2-B^2-A^3, and so on.

- As in simple A-B-A form, sectional balance is crucial if the form is to work well. Recurrences of section A, for example, need not be all of the same length; in fact, they should vary in interesting ways to avoid a feeling of outright repetition.
- The length of B, C, and D—or of as many new interludes as you care to include—will also vary, depending on the material you use. Either they will dictate their own length (longer for rich material, shorter for sketchy material), or you can experiment with mathematical proportions: gradually longer interludes (for development, or increased relaxation), or progressively shorter ones (to increase tension, impact, and momentum).

project 5

theme and variations

The theme-and-variations form consists of a principal section followed by a series of sections based on variations of the same material. In this case, there are no contrasting interludes: theme, variation 1, variation 2, etc., are all related and follow consecutively.

Be sure to establish the theme at sufficient length, and with a clear enough character, so that your audience can relate the variations to it.

In classical music, variations are generally as long as the theme itself. You can follow this proportion or adapt length to the material. If, for example, one section is built on faster speed variations, you may have said all you need to say in half the time; or a variation may be longer than the theme if it is built on long tones, slow speeds, silences, or reverb effects that need time to grow and recede.

A piece like this is ideal for exploring a wide range of tape and/or electronic modifications—provided that you adhere to the spirit of the form.

- Establish the theme.
- Give each variation its own character.
- Develop character by limiting each variation to a small number of modification techniques. (Throwing in a whole bag of sonic tricks before the finale may make an exciting section, but leaves nothing for the rest of the piece.)
- Plan out the sequence of variations so that the piece hangs together. This means maintaining both unity and continuous interest, perhaps through variations of progressive complexity.

project 6

a piece of measured lengths

The principle of this form is mathematical progression: a succession of events which become either progressively longer, progressively shorter, or some combination of the two. This can be done—using a ruler to measure tape lengths, or a stopwatch to time durations—by working with fixed proportional units.

- Each new section can be a certain number of seconds longer (or shorter) than the preceding one.
- Each new tape segment can be a certain number of inches longer (or shorter) than the preceding one.
- Sections can be alternately longer and shorter.
- Lengths can be increased (or decreased) in the first half of the piece, then reversed in the second half: an arch form.

If your materials, structuring, and proportions are just right, the form can produce dramatic momentum, a sense of breadth and expansion, or an intricate interplay between expansion and compression.

sounds in informal structures

project 7

a mood piece

The essence of a mood piece is its capacity to evoke a specific emotional response in the listener:

- agitation, anguish, celebration, comedy, drama, ecstasy, elation, enchantment, exuberance, fright, fury, gaiety, horror, impending doom, meditation, mystery, pain, passion, peacefulness, playfulness, ordeal, resignation, savagery, solemnity, tenderness, tension, weariness.

First, choose material and modifications that seem to evoke the desired response; second, develop the material but reject anything that deviates from a consistent mood; and third, try to sense how far you can go without overstating the point of the piece, or lapsing into useless repetition.

project 8

a descriptive piece

The essence of a descriptive piece is its capacity to evoke a specific image:

- children at play, a tidal wave, wind in the trees, a battle, an ancient legend, the seasons, a painting, a landscape, machinery, a celebration, outer space.

The difficult element is the selection of the mood to evoke the image. Mental imagery is highly subjective (sounds of war in your mind may conjure up an earthquake in your audience's).

Obvious images are easier to evoke: for instance, we frequently associate outer space with eerie sounds and strange silences. But subtler images—like the four seasons, a landscape, a myth—are more difficult. Choose not only your materials and modifications with care, but your descriptive title as well.

Listen to the way similar problems have been solved by other composers before attempting anything too subtle.

Bartók: *The Miraculous Mandarin*
Bernstein: *On the Waterfront*
Carlos: *Sonic Seasonings*
Colgrass: *As Quiet As . . .*
Copland: *Appalachian Spring*
Dodge: *Earth's Magnetic Field*
Gould: *Fall River Legend*
Holst: *The Planets*
Ives: *Fourth of July*
Ligeti: *Atmosphères*
Messiaen: *Couleurs de la cité céleste*
Mimaroglu: *Bowery Bum*
Mussorgsky: *Pictures at an Exhibition*
Ravel: *Jeux d'eau*
Sibelius: *Swan of Tuonela*
Stravinsky: *Petrouchka*
Subotnick: *The Wild Bull*
Villa-Lobos: *Forest of the Amazon*

project 9

an abstract work

A freeform composition is probably the most fun, and the most difficult—and for the same reason. Its shape is indefinite, there are no precedents for proportion, sequence, or duration, and no preconceived elements (repetition, contrast, variation, mood, images) to guide either the spirit of the piece or its structure.

You are dealing, instead, with one or another of the elements of sound itself: line, color, texture, spatial movement, register, speed, envelope.

The simplest way to work is to select an element, or an interplay of elements, as the basis for the piece. If you want to explore timbre, then make timbre modifications the chief experimental area until you have exhausted its possibilities. If you want an interplay of speed and envelope, of color and texture, or of any combination of sound parameters, then make those elements the object of exploration.

The proportional or sectional aspects of the piece, and its overall shape and duration, are usually determined during the development of the material. A logical splicing may be suggested, an interesting progression of modifications or a particularly dramatic or effective sequence may evolve, and so on. If nothing suggests itself, then review your objectives (What is the piece supposed to do?), and try out random mixtures and sequences. No matter how arbitrary the procedure may seem, it will at least produce something to accept or react against as a beginning.

sounds to accompany another medium

project 10

music to accompany a light or slide show

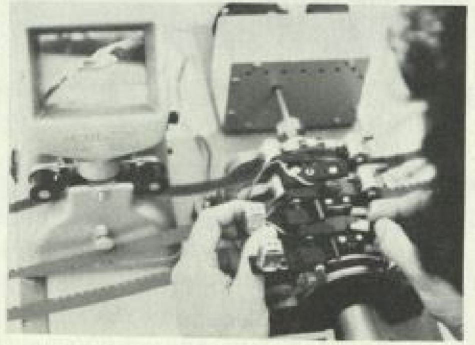
You can create electronic music for your own light or slide show, or for one produced with friends, the school art department, or a camera club. First, plan the mood, environment, character of visual events, pacing, and the possible addition of sound effects, narration, and other music—live or recorded.

Next, plan the sequence of visual events. Identify the visual characteristics of each event: color, mood, rhythm, pacing, image identity (if any), etc. Use contrast, variation, and repetition to develop audio elements that relate to the visual elements.

- A fast-moving visual sequence can be accompanied by fast-moving sound (repetition), or by slow-moving sound (contrast).
- The mood and color of the audio events can either duplicate or contrast with the mood and color of the visual events.
- An image of a sound-producing object can be accompanied by the actual sound it produces (repetition), an abstraction of the sound (variation), or a totally different sound (contrast).
- A close-up image can be accompanied by a close-up sound, or contrasted with a distant sound—or vice versa.
- A sound can be used to create an off-screen image that will enrich or even change the meaning of a visual image. The presence of an off-screen monster, for example, might change the meaning of an otherwise ordinary scene. When sound accompanies image, the two tend to be perceived as a single audio-visual event: the sound can reinforce or alter the perception of the image, while the image can reinforce or alter the perception of the sound.
- Junctures between sounds can be synchronized with junctures between images—or the sound can change while the image continues, and then the image change while the sound continues.

Carefully structuring these relationships can produce an integrated audio-visual composition, rather than just sounds that seem to tag along with images.

project 11



film scoring

The problems encountered in scoring a film are basically similar to those of slide- or light-show scoring. In this medium, however, the technical aspect of syncing audio-visual events is particularly crucial. To score a film effectively, you have to know the precise length of each visual event. If you know the footage or number of frames in each scene you can calculate exact projection times. The footage or frame-count—which you will need for stopwatch-timing of your music—is contained on a cue sheet which a professional producer will provide. If you are working with an amateur group, you may have to sit down with the film editor to make your own cue sheet.

There is a great deal to be said about film scoring which is beyond the scope of this book. Your best bet, before you start, is to learn as much as you can from books such as *Scoring for Film* (Hagen), *Underscore* (Skinner), *Sounds and Scores* (Mancini), and *Music in Modern Media* (Dolan).

project 12

music for dance

Unless you have had some personal experience in planning dance steps, movements, and staged patterns, you should consult an experienced dancer or choreographer before attempting a dance score.

Dance music has its own peculiarities, based on the fact that the performers move on stage in coordination with the sounds they hear. If the music is rhythmically vague or texturally monotonous, then the dancer has nothing with which to synchronize his movements. This does not mean that your music has to be filled with drum beats, or even structured in regular meters. What is required is an overall feeling of movement, with clearly marked sections and easily discernible events so that the dancers will know where they are.

project 13

words and music

One of the most interesting projects for the electronic music composer (or any composer) is the creation of sounds to accompany the spoken word. The verbal part—a poetry reading, stage play, improvised talk, vocal sounds, or material taped from records, radio, or TV—may be unaltered or modified by tape techniques and incorporated in the music.

To create sounds for a reading or stage production, begin by answering such questions as:

- What is the mood of the words?
- How and when does the mood change?
- What is the pacing of the words (or the tempo of the stage action)?
- Does the text need a mood-setting introduction? a conclusion?
- Are certain sections more effective without music?

- Are there staging gaps, with no one speaking, when continuity would depend on music?
- Are there silences in the speech that need music in the foreground?

If, on the other hand, you intend to use speech or vocal sound as an integral part of a musical complex, treat it as a sound to be modified like any other.

Voice sounds frequently carry emotional overtones and evoke specific images. For example, the words "power," "death," and "magic" carry connotations which can add dimension to a piece. A sigh, heavy breathing, or an echoing scream can call up strong images in the listener's mind.

A nonsense word—a meaningless, made-up sound—can be funny, threatening, or mysterious, provided that we modify the sound accordingly.

Foreign-language words, as sounds, can also be suggestive. Think of your reaction to soft words like "sempre caro" or "les violons d'autan," harsh sounds like "achtung" or "sehr gut," the rhythm of Chinese, or the sonorities of Spanish or Russian. (Check out the literal meaning in case the effect might be altered for the listener who understands the language.)

For examples of this type of work, listen to some of the following:

Ashley: *She Was a Visitor*
Babbitt: *Vision and Prayer*
Berio: *Visage*
Omaggio a Joyce
Cage: *Solo for Voice 2*
El-Dabh: *Leiyla and the Poet*
Feldman: *Chorus and Instruments (II)*
Gaburo: *Antiphonie III*
Oliveros: *Sound Patterns*
Reich: *It's Gonna Rain*
Stockhausen: *Gesang der Jünglinge*
Takemitsu: *Vocalism Ai (Love)*

project 14

mixed-media productions

Contemporary adventures in mixing media—combining different art forms in the same work are neither new nor revolutionary. Nineteenth-century visionaries like Richard Wagner and Franz Liszt spoke of an "artwork of the future" which would combine many, if not all, arts in a single work; and Alexander Scriabin, a composer-mystic of the early twentieth century, was obsessed with the achievement (never realized) of a colossal and "perfect synthesis of the arts" to be used in the service of religion.

What we have today—if not an original idea—is an intensification and expansion of the mixed-media concept due both to the expanding possibilities of our technology and to a search for new means of expression.

With tape techniques and electronic sound at your fingertips; with sophisticated lighting and staging technology; with creative people around you in art, film, theater, dance, and musical performance—there are endless artistic avenues to be explored.

Mixed-media productions, like those of dance and theater, are essentially group activities. Though music composition follows the same basic process whether done as an individual or within a group, the difference is that your musical creativity interacts with other art forms. This interaction can be a stimulating source of growth.

technical information

tape transport and tape heads

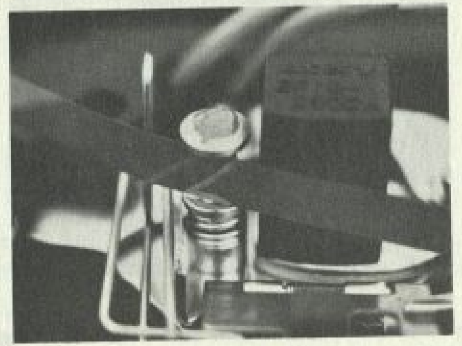
left spindle/feed (supply) reel

When the tape moves from left to right ('forward', 'fast forward', and 'record'), the left spindle holds the feed reel, maintaining proper tension on the tape. During rewind, the left spindle turns the feed reel clockwise, pulling the tape backwards past the heads.

right spindle/take-up reel

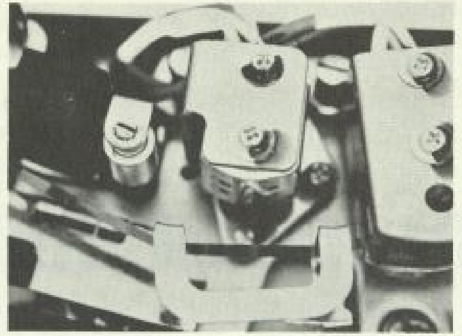
In any forward tape movement, the right spindle turns the take-up reel counterclockwise, moving the tape from left to right across the heads. During rewind, the right spindle holds the take-up reel, maintaining proper tension on the tape.

tape guides



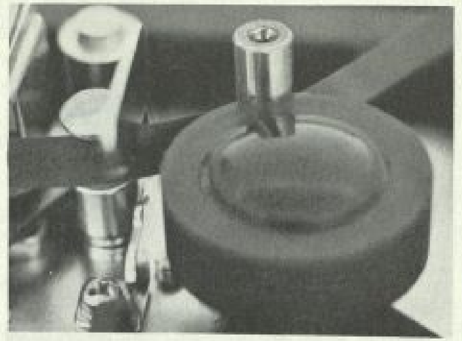
The guides are stationary posts that align the tape for travel past the heads.

tape lifters

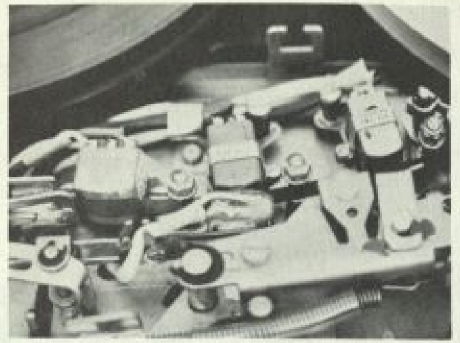


The lifters reduce wear on both the tape and the heads by separating them during 'fast forward' and 'rewind'.

capstan/pinch roller



The capstan is a motorized metal rod that works together with the pinch roller—a movable rubber wheel—to pull the tape past the heads.



erase head *record head* *playback head*

erase head

This head erases any previously recorded sound when the recorder is in 'record'. In both two- and three-head machines, it is located to the left of the other heads.

record head

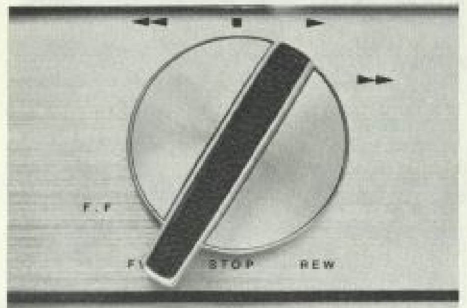
This head records sound by imprinting a signal on the tape in the form of a magnetized path or track. In a three-head machine, it is located in the center. In a two-head machine, record and playback functions are performed by the same head, located to the right of the erase head.

playback head

The playback head reads the magnetic patterns on the passing tape. These patterns are then amplified for conversion to an audible signal. In a three-head machine, the playback head is located to the right of the other heads. In a two-head machine, record and playback functions are performed by the same head.

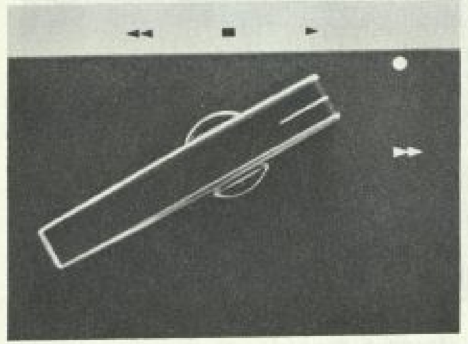
tape recorder controls

control knob



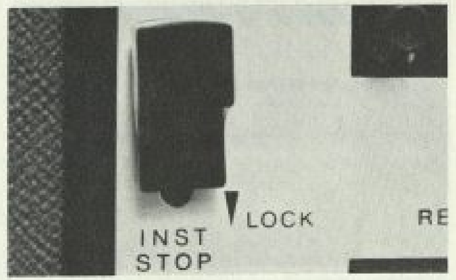
The control knob regulates tape movement in four ways:

- stop
- forward (or play)
- fast forward
- rewind



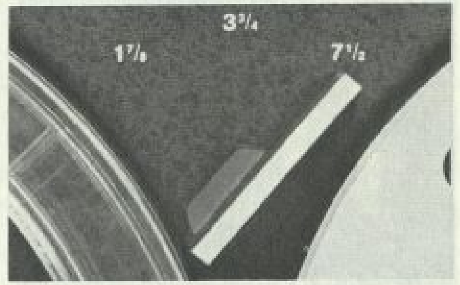
Some recorder controls incorporate a fifth position for pause control (instant stop).

pause control/instant stop



The pause control instantly stops a moving tape during recording or playback. When the control is activated, the record or playback head remains live, and the tape is freed from the capstan. This allows the tape to be moved by hand to locate a specific sound for editing or to produce manual speed change effects.

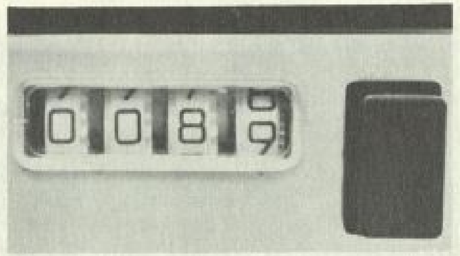
speed selector



This lever regulates tape speed. The choice of speeds—measured in inches per second (ips)—depends on the tape recorder model:

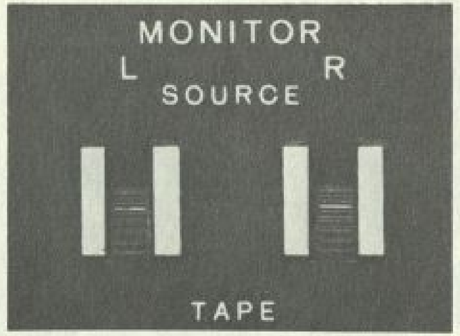
- 1 7/8 ips
- 3¾ ips
- 7½ ips
- 15 ips (professional models)

tape counter



The tape counter measures reel revolutions by number: from '000' to '999' or from '0000' to '9999', depending on the recorder model. A button or wheel, located next to the indicator, is used to re-set the numbers to zero for each new tape count.

monitor switch



A monitor switch is found only on three-head recorders. In 'source' position, it allows the operator to monitor incoming signals. In 'tape' position, it allows the operator to monitor recorded signals. Switched back and forth from 'source' to 'tape' during recording, it allows the operator to compare incoming with recorded signals (A-test/B-test).

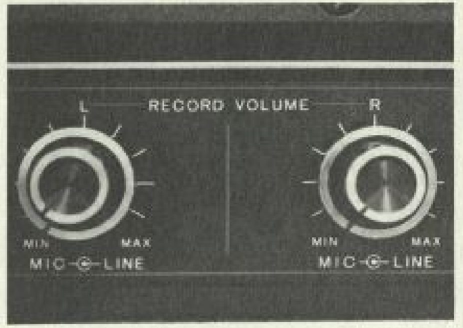
inputs and outputs

Input jacks accept mic and line plugs for recording. Output jacks provide the means to patch the output signal to another component, an amplifier, a speaker, another recorder, a synthesizer, etc.

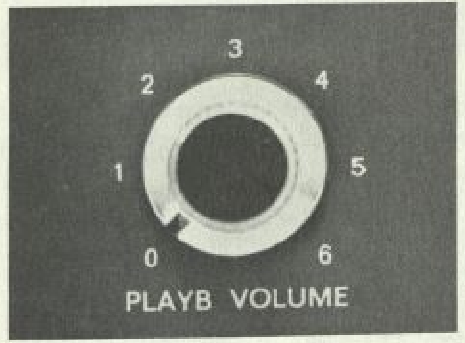
record button(s)

Pressing the record button(s) activates the record head to imprint any incoming signals on tape. (Check the manufacturer's manual for specific procedures for your recorder.)

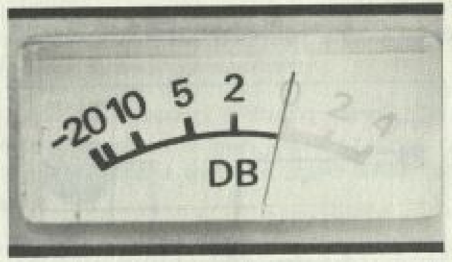
volume controls

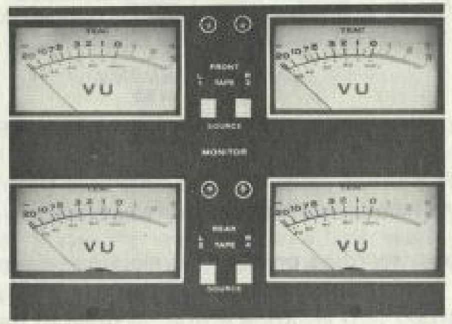


One kind of volume control regulates recording level—that is, the signal strength during recording. The other regulates playback level—the signal strength passing from the recorded tape to the amplifier and loudspeaker.



VU meter(s)





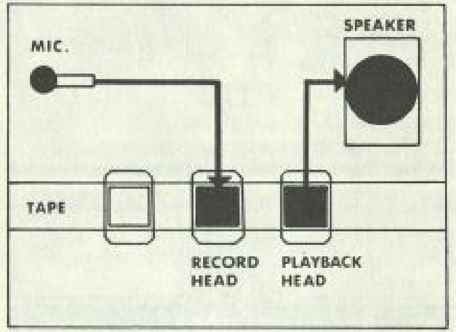
The VU (volume unit) meter provides a visual indication of the loudness or decibel (dB) rating of the program being recorded or played back. Any change in performance volume, level control setting, or source/mic distance is immediately indicated by a change in the position of the needle (or other indicator).

A reading at the "0" level (which does not mean "zero dB") indicates that the signal has reached its maximum strength for recording or playback without distortion. Constant recording or playback in the red area above "0" creates distortion; constant recording or playback in the lower black area—well below "0"—produces faint or inaudible sound.

The number of VU meters depends on the number of channels: mono instruments have one; stereo, two; quad, four.

channels, signals, and tracks

the channel



A channel is a single path of sound that uses electronic equipment to travel from a sound source to a loudspeaker.

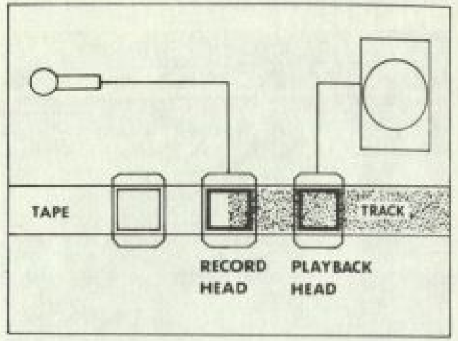
Since the tape-recording process uses electrical energy for every component from microphone to loudspeaker, the sound itself must be changed into electrical energy in order to travel through a recorder's circuitry.

the signal

When a sound is fed into a recorder through a microphone, the microphone converts the vibrations of sound into corresponding fluctuations of electrical current: a signal.

When a sound is fed into a recorder from an electronic source (a radio, a phonograph, another tape recorder, a synthesizer, an electric guitar, etc.), the incoming sound is already in the form of a signal. No conversion is necessary. The signal simply passes from the line output of one electronic instrument (the sound source) to the line input of the other (the recorder).

the track



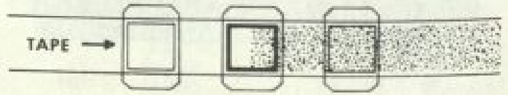
As the signal passes through the record head's electromagnet, all of the electrical impulses (the equivalent of the sound vibrations) are imprinted on the passing magnetic tape. This imprint is in the form of a magnetized path or track.

The number of tracks that can be imprinted on a tape depends on two things: track width and tape width.

track width

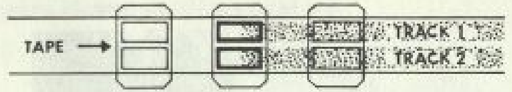
The width of a track depends on the size of the electromagnet in the record head that sends the signal to the passing tape.

If the head consists of a single magnet that covers almost the full width of the tape, it will imprint its signal on a single track of the same width.



This is called full-track recording.

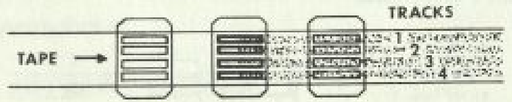
If, in place of one full-size magnet, the record head has two half-size magnets (one above the other), the two will imprint two tracks on the tape (one above the other).



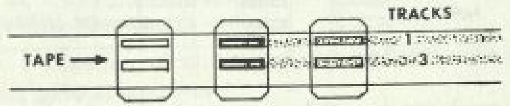
This is called half-track or two-track recording.

To make a quarter-track or four track recording, there are two possible record-head formats.

- For quadraphonic recording, the head contains four quarter-size magnets (stacked one above another) which imprint four tracks on the tape.



- For quarter-track stereo recording, the head contains only two quarter-size magnets positioned in this way:



(How these two magnets imprint four tracks is described on p. 88.)

signal-to-noise ratio

In the recording process, the strength of the signal on a track is always compared to the strength of unavoidable noise (such as tape hiss) in the recording equipment.

The most desirable situation—a strong signal and a minimum of noise—is described as a high signal-to-noise ratio. A low signal-to-noise ratio means that the signal tends to be obscured by the noise.

Because of its larger signal area, a wide-track recording has a higher signal-to-noise ratio than a narrow-track recording.

crosstalk

Another source of unwanted sound is signal leakage (crosstalk) from one track to another during playback. Although a certain amount of crosstalk is to be expected on multi-track tape (at an extremely low level), the degree of interference increases when a number of tracks are imprinted on a tape, leaving little room between them. Fewer tracks, or more space between adjacent tracks, provide a cleaner sound.

tape width

To produce the best possible recording (high signal-to-noise ratio, minimum crosstalk), professional tape studios adjust tape width to the recording situation.

- conventional quarter-inch tape for full-track and half-track recording
- half-inch tape for four-track recording
- one-inch tape for eight tracks
- two-inch tape for sixteen tracks

However, if tape versatility (a greater number of available tracks) is more important than high sound quality—which is the

case in home recording equipment—then both the width of the tape and the size of the record head can be reduced.

Three examples of such recording formats are the stereo cassette (four tracks on eighth-inch tape), quadraphonic reel-to-reel (four tracks on quarter-inch tape), and the eight-track cartridge (on quarter-inch tape).

two-directional tracks

The familiar and economical way to use a tape cassette is to record side A, then (without rewinding) to turn over the cassette to record side B. This means that you are recording on only half of the tape width at a time.

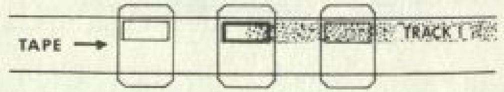
- 1st recording: signals are imprinted as a track running along the top half of the tape.
- Cassette turned over: the recorded track (now running backwards) is on the bottom; the blank half is on top.
- 2nd recording: signals are imprinted as a track running along the new top half of the tape.
- Result: one tape with two tracks running in opposite directions.
- Playback: each track is heard separately, played back in the same direction it was recorded.

The same procedure is also used with certain formats of reel-to-reel recorders—and for the same reason: to double the amount of available recording time on a reel of tape.

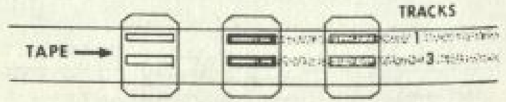
In this case, the reels are lifted from the spindles after the first track is recorded (without rewinding the tape), then turned over and replaced on the opposite spindles: original feed reel on take-up spindle, original take-up reel on feed spindle.

The two most common examples of this format are:

- half-track mono



- quarter-track stereo



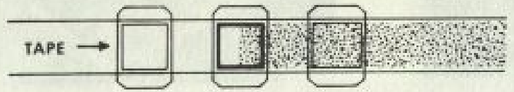
In quarter-track stereo, all four tracks are imprinted by only two quarter-size magnets. After tracks 1 and 3 are recorded, the tape is turned over, and the same magnets then imprint tracks 2 and 4 in the blank areas.

one-directional tracks

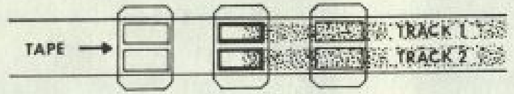
In contrast to a two-directional format, recorders with one-directional formats record and play back all tracks in the same direction.

Three common examples of this type are:

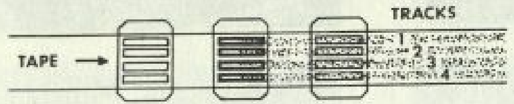
- full-track mono



- half-track stereo



- quadraphonic



which system to use

If there is a question of choice among available tape recorders, follow these general guidelines:

- One-directional recording is essential if the tape is to be edited. If tracks have been recorded in both directions, it is impossible to edit one without ruining the other.
- Two-directional recording is a tape-saver. It doubles the amount of recording time per reel.
- Wide-track formats produce superior sound. They are ideal for original recordings in mono or stereo.
- Multi-track formats provide channel separation for multi-layered sounds. They are used either for finished tapes, or for original tapes that require volume-controlled mix-downs to be made to mono or stereo.

number and nature of sources

The simplest kind of recording uses one mic or line input to carry one channel of sound to imprint one track. The signals on that track may originate as one sound source or as a number of sound sources combined by a "Y" connector or a mixer. If more than one channel is available (two for stereo recording, four for quadraphonic), sound sources may be separately tracked for clarification of sound layers.

Any tape recording—regardless of the number of channels used—may consist of signals from one or more mic inputs, one or more line inputs, or any combination of mic and line inputs.

simultaneous recording

A number of sound sources, occurring simultaneously, can be recorded on the same tape at the same time.

consecutive recording

A number of sound sources can be recorded consecutively, yet can be combined on the same tape for simultaneous playback.

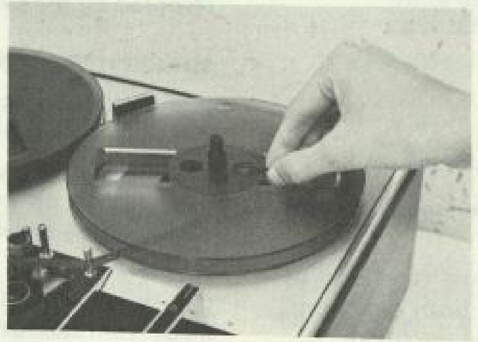
This may take the form of sound-on-sound recording (two or more signals on the same track), or sound-with-sound recording (two or more signals on different tracks).

Any signal added to an existing track may originate as a live mic input, a live line input, or a previously recorded track (line input from another tape recorder).

how to thread tape

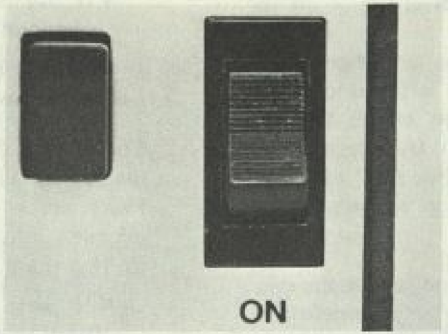
1. Place a reel of tape on the left spindle with the tape end hanging out on the left side. This is the feed or supply reel.
2. Place an empty reel on the right spindle. This is the take-up reel.
3. Gently pull a few feet of tape from the feed reel, drawing it
   - in front of the first tape guide;
   - past the tape heads;
   - between the capstan and the pinch roller, then past the second tape guide.
4. Bring the tape end up to the right side of the take-up reel and insert it in the slot of the reel hub.
5. Gently turn the take-up reel in a counterclockwise direction until the tape is firmly secured around the hub. If necessary, turn the feed reel at the same time (also counterclockwise) to relieve tension on the tape.

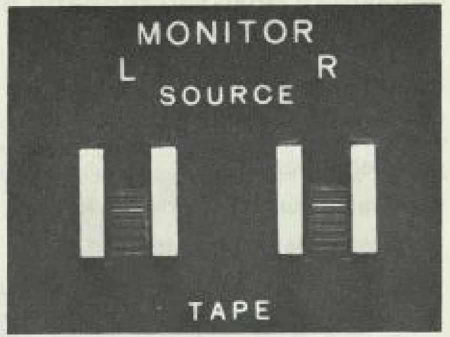
Before recording or playback, make sure that there are no twists in the tape and that its magnetic side is facing the tape heads.

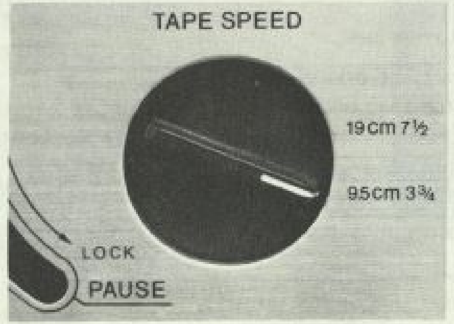


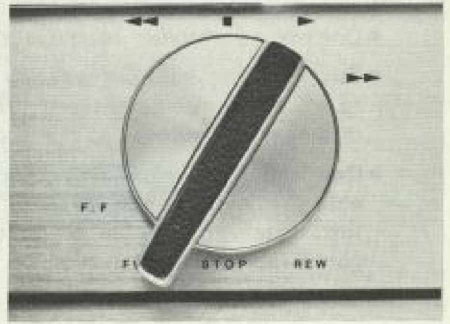
playback procedure

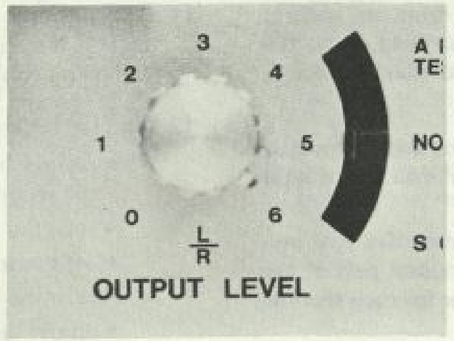
1. Power on.
2. Tape threaded. For immediate playback after a recording, rewind to zero (or the first tape counter number for the recording).
3. Monitor switch on 'tape'.
4. Speed selector set.
5. Control knob in 'forward' (or 'play') position.
6. Volume control set for the best playback level.

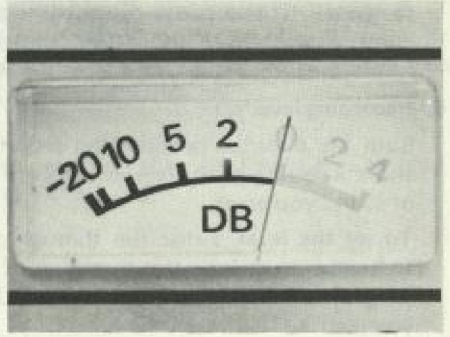












recording with microphone inputs

1. Power on.
2. Tape threaded.
3. Monitor switch on 'source'.
4. Speed selector set.
5. Tape counter indicator set at zero.
6. Microphone(s) connected.

   A mono recorder has a single mic input jack to receive the mic plug. Stereo has two, marked 'L' and 'R' (left channel/right channel). Quad has four, marked for left, right, front, and back channels.

   - One mic per channel: insert one plug into one jack.
   - Two or more mics per channel: connect the mic plugs to a "Y" connector or mixer; then patch the connector or the mixer's output into the recorder jack.
   - One mic for two or more channels: connect the mic plug to the single arm of a "Y" or multiple connector; then patch the double or multiple arms into the recorder jacks—one per channel.

7. Machine in 'record'.

   Hold down the record button(s) and turn the control knob to 'pause'. If the pause control is separate from the control knob, first activate the pause control, then hold down the record button(s) and turn the control knob to 'play'.

8. Recording level set.

   Both the distance between the sound source and the mic(s), and the setting of the level control are determined by the kind of sound you want.

   To set the level, either run through an unrecorded trial performance or make a test recording of the loudest part of the program. As you do so, watch the VU meter for each channel you use.

   - If the needle remains in the red area, the level is too high and recording will be distorted.
   - If the needle remains in the lower black area, the level is too low and recording will be faint or inaudible.

   To achieve the best recording level—generally in the upper black area, with occasional peaks into the red and dips into the lower black—make the appropriate adjustments: increase or decrease mic/source distance; raise or lower the level setting, or, if you are using a mixer, adjust its level controls.

   - If parts of the program are too low as a result of the best level setting, or if the program is such that it can't be tested in advance, you will have to "ride gain"—that is, make appropriate control-setting changes while the performance is in progress.

9. Control knob

   Turn the control knob to 'forward' (or 'play') position.

10. To stop recording, either
    - turn the control knob to 'stop' position (for a full stop between takes); or
    - activate the pause control or instant stop (for a short interruption in recording).
11. To continue recording after a full stop, repeat steps 7, 8, and 9. To continue recording after an instant stop, release the pause control.
12. To play back the recording,
    - rewind to zero (or the first tape counter number for the recording),
    - turn the monitor switch to 'tape',
    - set the speed selector for playback speed,
    - turn the control knob to 'forward',
    - adjust the playback level.

recording with line inputs

(Direct-wire recording of electronic instruments)

1. Connections

   A mono recorder has a single line input jack to receive the 'line out' plug from an electronic instrument or its amplifier. Stereo has two, marked 'L' and 'R' (left channel/right channel). Quad has four, marked for left, right, front, and back channels.

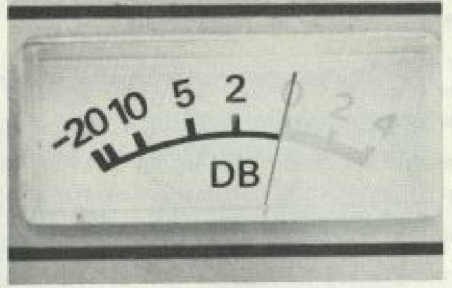
   Before patching the electronic instrument into the recorder's 'line in' jack, you will have to determine the strength of its signal.

   Instruments with a high-level signal (radio, electric guitar, amplifier, etc.) can be patched in directly. Those with a low-level signal (phono cartridge, electric guitar pickup) require pre-amplification to raise signal strength to a usable level.

   Two or more instruments patched into the same channel must have compatible signals. For example, you can mix the phono cartridge and the guitar pickup (both are low), but not the cartridge and the guitar amplifier (one low, one high).

2. Power on (all equipment).
3. Tape threaded.
4. Monitor switch on 'source'.
5. Speed selector set.
6. Tape counter indicator set at zero.
7. Record button(s) pressed with control knob on 'pause'—or on 'play' with separate pause control activated.
8. Recording level set.

   Sample the sound to be recorded through either an unrecorded trial performance or a test recording of the loudest part of the program. Set the instrument's amplifier at the highest level that does not produce distortion or amplifier noise. This will give you the best signal-to-noise ratio. As you sample the sound, watch the VU meter for each channel you use.



   To achieve the best recording level—generally in the upper black area, with occasional peaks into the red—make the appropriate volume adjustments, depending on the components you are using—the recorder itself, plus any pre-amps, amps, or mixers. If parts of the program are too low as a result of the best level setting, or if the program is such that it can't be tested in advance, you will have to ride gain.

9. To start recording, to stop and then continue, and to play-back, follow steps 9 to 12 on p. 92.

tape editing procedures

direct-wire dubbing

1. Connections
   Patch Recorder 1's line output to Recorder 2's line input.
2. Power on, both recorders.
3. Tape
   Recorder 1: thread the recorded tape.
   Recorder 2: thread a reel of blank tape.
4. Monitor switch
   Recorder 1: on 'tape'.
   Recorder 2: on 'source'.
5. Speed selector set, both recorders
   (same or different speeds, depending on the effect you want).
6. Tape counter indicator set at zero, both recorders.
7. Record button pressed, Recorder 2 only.
8. Recording level set during trial playback of Recorder 1. Watch VU meters; adjust #1, then #2. Rewind recorded tape to zero.
9. Recording
   Recorder 2: switch control knob to 'forward'.
   Recorder 1: switch control knob to 'forward'.

simultaneous editing and dubbing

During the dubbing operation outlined above, portions of the original (Tape 1) can be omitted on the new recording (Tape 2) by activating different controls.

As a result of this start-stop technique, sounds on the receiving tape come out in edited form while keeping the original tape intact.

- **Feeding tape stopped/receiving tape running**
  When Recorder 1's pause control is activated, or its control knob switched to 'stop', the feeding tape stops instantly for an abrupt cut-off of the sound source. The receiving tape records silence until the pause control is released, or the control knob is returned to 'forward'.

- **Feeding tape running/receiving tape stopped**
  When Recorder 2's pause control is activated, or its control knob switched to 'stop', the recording process stops instantly. Since the feeding tape runs without interruption, the recording can be resumed by releasing the pause control. This editing method produces a dub with no silences between the recorded takes.

- **Editing while both tapes are running**
  Without interrupting tape movement on either recorder, portions of the original taped sounds can be omitted on the new recording by activating Recorder 2's muting control.

  This device—available on some models—momentarily cuts off the sound input while recording is in process. The result is a dub with silence between each take.

  Finally, sounds can be deleted abruptly or gradually (fade out/fade in) by manipulating Recorder 2's level control during the dubbing.

- **Dubbing different tapes in sequence**
  An uninterrupted series of different sound events can be dubbed on the same blank reel by stopping both recorders, changing source tapes (Recorder 1), and then resuming the recording.

  If this procedure is carefully planned and controlled, any number of sound events can be linked together to form part of a longer work or even the complete work itself.

cutting and splicing

Cutting and splicing, a simple, fundamental technique, is used by composers, editors, and recording engineers to obtain precise, noise-free connections between different tape segments.

The technique can be used alone (a tape processed entirely by cutting and splicing) or in combination with the editing-dubbing techniques described above. In either case, its purpose is to arrange or rearrange sound events in sequence, and to delete unwanted sounds and silences.

Equipment

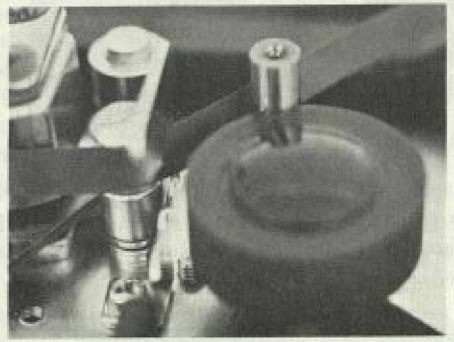
- splicing (editing) block: a metal cutting surface with a tape-holding channel and one or two razor grooves
- small, empty reels (3", 5" size) to store labeled tape segments
- grease pencil to mark the tape
- single-edge razor blade
- splicing tape (with dispenser)
- leader tape

Preparation

- Clean all equipment. Dirt and grease prevent secure, noise-free splices.
- Demagnetize the razor (or use a new blade). A demagnetizer (degausser) is an inexpensive but essential piece of equipment for the tape studio.
- Choose splicing tape that is slightly narrower than the recording tape. Wider splicing tape "bleeds," sticks, and collects dirt.
- Lay out tools, materials, recorders, and any accessories in order and within easy reach. Set up the roll of leader tape on a spindle or dispenser for quick access.

Procedure

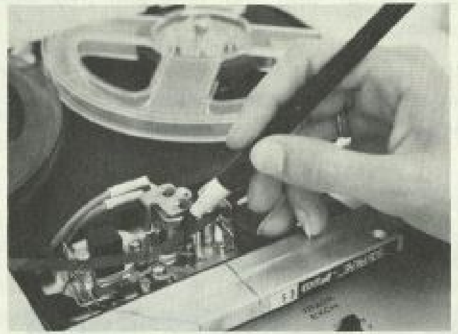
1. Power on.
2. Thread and play the tape normally. When it reaches the approximate editing point, either
   - press the 'stop' or pause button (if your recorder has this feature) to stop the tape and free the transport mechanism, or
   - stop the tape, using the control knob, then by-pass the capstan so that the tape passes directly from the heads to the take-up reel.



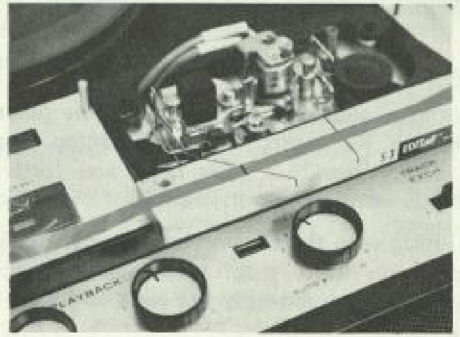
3. In either case, the reels can now be turned by hand.
   - If you have used the pause button, the playback head is still live. To locate the exact editing point on the tape, slowly turn the reels back and forth until the sound you want crosses the center of the playback head.
   - If you have used the capstan-by-pass method, switch the control knob from 'stop' to 'forward' to activate the playback head. Hand-turn the reels to locate the exact editing point.

Note that the slow, irregular tape speed produced by hand-turning the reels will considerably alter the pitch of the recording.

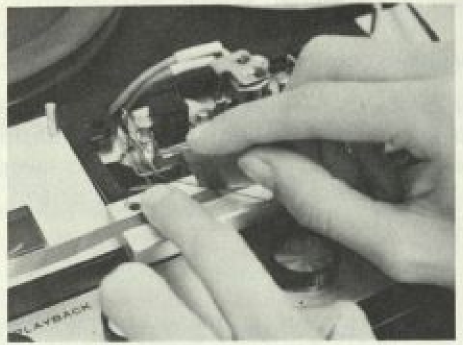
4. Marking is done with a grease pencil on the side of the tape *away* from the heads. Draw a single vertical line where the chosen sound reaches the center of the playback head.



5. Lift the tape away from the heads and place it, marked side up, in the splicing block's tape channel. Position the tape so that the center of the grease pencil mark crosses the diagonal razor guide.

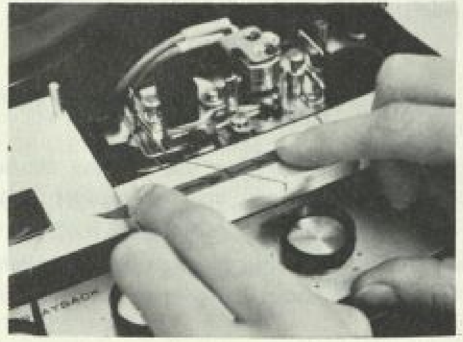


6. Cut the tape by sliding the razor blade through the guide.

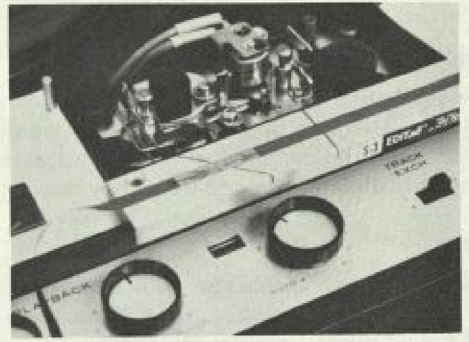


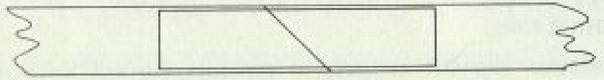
7. Locate the sound to be joined to this one, and mark and cut the tape.

8. Place both tapes in the channel. Butt the ends, avoiding either a gap or an overlap.



9. Lay a short piece of splicing tape across the joint and rub it down with your fingernail.





An angled cut, and splicing tape in the position and of the approximate length shown in the illustration, will create a smooth transition.

splicing sound-to-sound, sound-to-silence

Using the same editing technique—locating and marking a sound, cutting and joining the tape ends—a complete composition can be built up by sequencing any number of sound events in any order:

- different portions of the same source tape;
- portions of different source tapes;
- added silence before, during, or after a sound event.

General guidelines

- Cut as close as possible to the beginning of the chosen sound. This eliminates unwanted sound or tape hiss before the attack.
- Cut as close as possible to the end of the chosen sound to eliminate unwanted sound or tape hiss.
- Cutting too close to the beginning or end of a sound may kill its attack or shorten its decay. This, however, may be done intentionally to modify the sound.
- To add silence before, during, or after an event, splice in a piece of leader; the length of silence desired determines the length of leader.
  Paper leader produces total silence since it has no recording properties; plastic leader may carry a slight amount of static, audible during playback.
  Blank recording tape, used as leader, produces some degree of tape hiss.
  Recorded silence (room sound) can be used to separate two sounds, while retaining the feeling of their environment. The naturalness of the silent bridge overshadows unavoidable tape hiss. (Room sound is recorded in silence, with the same equipment and in the same area used for the sound recording.)
- For intricate cutting involving several short tape segments, make identifying marks with grease pencil on the back side of each piece: code numbers or letters which match a written cue sheet; arrows showing the direction of tape travel; one vertical mark for the beginning of a sound, and two marks for its end, etc.

adding leader

For easier threading and handling, a few feet of leader should be added to both the head and the tail of the tape. Splice the head leader to the beginning of the first attack, and the tail leader to the end of the final decay.

microphones

Microphones are categorized by their directional sensitivity and by the type of system used to convert sound waves to electrical current. Their recording characteristics are determined by their range of frequency response and by their sensitivity to different frequencies within this range. Relatively equal sensitivity, termed flat response, produces minimum distortion.

directional characteristics

omnidirectional

An omnidirectional microphone has relatively equal sensitivity to sound coming from all directions. Sounds, consequently, cannot be selectively recorded. This characteristic is generally undesirable for electronic music recording.

bidirectional

A bidirectional microphone has maximum sensitivity to sounds coming from two opposing directions. This characteristic, useful for recording dialogue, is generally undesirable for electronic music.

unidirectional or cardioid

A unidirectional microphone has maximum sensitivity to sound coming from one direction and less sensitivity to sound coming from behind it. The resulting heart-shaped field pattern is termed cardioid. It is the most useful microphone for electronic music recording since its directional characteristic makes it possible to record with a high degree of selectivity.

superdirectional

Superdirectional, or shotgun, microphones have extremely directional characteristics, but many have a more limited frequency range than cardioid microphones.

type

carbon mic

The changing pressure of sound waves produces current by causing variations in resistance between encapsulated carbon granules. This type of microphone, used in telephones, is extremely inexpensive but has the most limited frequency response.

crystal mic

Sound waves move a diaphragm which causes an attached crystal to react by generating current. This type is also inexpensive and has a limited frequency response.

ceramic mic

A ceramic element is used to generate current by a method similar to that used in crystal mics. Though inexpensive, this type has a wider frequency range and flatter response.

dynamic mic

Sound waves move a diaphragm which in turn moves a coil back and forth within a magnetic field. The frequency response of this type is relatively wide and flat. It is in the medium price range. For these reasons it is most frequently used for quality home recording.

condenser mic

Sound waves move an electrically charged diaphragm suspended in front of an electrically charged plate to vary capacitance and generate current. As this type is capable of the widest and flattest frequency response, it is used almost exclusively in professional recording. Most condenser mics are extremely expensive, though some, made for amateur recording, are modestly priced.

glossary

additive synthesis Combining waveforms to create new waveforms.

ADSR Label abbreviation, used on an envelope generator, to identify the attack/decay/sustain/release envelope phases.

ambient noise Sounds that are part of, and natural to, a recording environment.

amplifier A device which increases the strength of a signal.

amplitude The relative height of a waveform, measured from the point of rest; a characteristic of sound which determines its loudness.

amplitude modulation (AM) A change in a signal's amplitude produced by a modulating or "program" waveform.

attack The beginning envelope phase of a sound as it rises to maximum amplitude.

audio generator (or oscillator) An electronic device that produces simple or complex waveforms in the audible range.

band-pass filter A filter that subtracts both the high and low frequencies and passes the center frequency.

band-reject filter A filter that subtracts or rejects the center frequencies and passes both the highs and lows.

bulk eraser A device that erases an entire reel of tape at one time.

canon A contrapuntal technique, like a round, in which a phrase of music is completely and almost immediately restated in a second voice.

carrier In frequency or amplitude modulation, the audio signal as opposed to the modulating signal.

channel A single path of sound that uses electronic equipment to travel from a sound source to a loudspeaker.

crosstalk Signal leakage from one track to another during tape playback.

cutoff point A filter setting which determines the line between passed and rejected frequencies.

cycle One complete vibration or oscillation of a waveform.

cycles per second Also designated as 'cps', or 'Hz' (hertz). The number of complete vibrations that occur in one second, determining the frequency of a sound.

decay 1) The final envelope phase as a sound declines to silence or 2) the second envelope phase as amplitude declines from attack peak to sustain level.

direct-wire recording Recording from an electronic source rather than a live one.

dub Make a copy of a recorded tape. Also used to refer to the copy itself.

echo In electronic music, the effect produced on a three-head tape recorder by re-recording the played-back signal.

edit In tape composition, to change the shape or sequence of recorded sound events by deletion, addition, or rearrangement.

electronic switch An electronic device that periodically interrupts a single signal or alternates two signals.

envelope The contour or shape of a waveform or the sound it produces, characterized by amplitude changes that determine its growth and decay.

envelope generator A control-voltage source, used in conjunction with a voltage-controlled amplifier, that shapes a sound envelope by regulating its attack/decay/sustain/release phases.

event The basic element of electronic music, in the form of a taped unit of sound, used as raw material for composition.

feedback The re-routing of a signal—resulting in a repetitive or echo effect—from the output of a tape recorder to its input or from a microphone to speaker.

filter A device that modifies sound by selectively subtracting and passing portions of the signal's frequencies.

fixed filter bank A filter that divides the entire audio frequency range into a number of bands which can be subtracted from the whole, either individually or in any combination.

frequency The number of cycles per second of a sound wave or signal. Designated as 'cps' or 'Hz'.

frequency modulation (FM) A change in a signal's frequency produced by a modulating or "program" waveform.

fundamental The lowest, or first, partial of a sound's frequency structure.

gain Degree of amplification.

glissando Gliding. A rapid slide, up or down, through consecutive pitches.

harmonic Also, overtone or partial. A single element of the frequencies present in complex sounds, mathematically related to the lowest partial, or fundamental, of that sound.

hertz Abbreviated 'Hz'. Cycles per second. Named for Heinrich Hertz, German scientist.

high-pass filter A filter that subtracts the frequencies below a given cutoff point, and passes those above.

impedance The resistance effect of a circuit or component; expressed in ohms.

input A signal fed into a circuit. Also, the jack or other connector for the incoming signal.

jack Receptacle for a plug connector leading to the input or output circuit of a tape recorder or other electronic device.

line A signal input (line in) or output (line out) for connecting electronic components.

linear controller *See* ribbon controller.

low-pass filter A filter that subtracts the frequencies above a given cutoff point, and passes those below.

magnetic tape A flexible, plastic ribbon for sound recording treated on one side with a permanent coating of magnetically sensitive metallic powder.

mic/line Circuitry which permits the simultaneous use of both a microphone and a line input.

mil A unit of measure, equal to one one-thousandth of an inch, used to gauge tape thickness.

mixer A device which combines two or more input signals for the purpose of feeding the combined signals to another audio component.

modifier An electronic component designed to alter a signal in a specific way.

modify In electronic music, to alter a signal through tape manipulation, tape editing, or by electronic means.

modulation Variation. Specifically, periodic variation of the frequency or amplitude of a waveform.

monitor To listen to a signal being recorded.

musique concrète Music that uses recorded sound sources other than electronic ones as raw material for composition.

noise 1) Unwanted sound. 2) A complex sound containing a broad spectrum of non-harmonic overtones.

oscillation Vibration.

oscillator An electronic device that produces one or more waveforms within a specific frequency range.

oscilloscope An electronic instrument that uses the screen of a cathode-ray tube to display a graphic representation of a signal in the shape of a waveform.

ostinato A clearly defined melodic or rhythmic pattern that is repeated persistently, usually in immediate succession.

output A signal coming out of a circuit. Also, the jack or other connector for the outcoming signal.

panning A recording term used to describe the panoramic movement of sound.

parameter An element that can be measured. Parameters of sound include frequency, timbre, amplitude, and envelope.

partial *See* harmonic.

patch cord A cord with plugs at both ends, jacks at both ends, or a plug and a jack at either end, used for connecting components.

peak The greatest amplitude of a sound wave or signal.

pitch The relative highness or lowness of a tone, determined by its frequency within the audible range.

portamento Also, glide control or glissando. An electronic keyboard control used to regulate varying degrees of glide between pitches.

potentiometer Also, 'pot'. A volume control on audio equipment.

pre-amplifier An electronic device that raises low-level signals to a level usable by other components.

print-through Also, layer-to-layer signal transfer. A sound-distortion factor caused by the transfer of magnetism from one layer of tape to an adjacent layer while the reel is in storage.

rectangular wave A signal having a rectangular waveform. The harmonic content varies with the proportion of the rectangle.

release The final envelope phase as amplitude declines to silence.

resonance control Also, regeneration control. A filter control used to emphasize the overtones closest to the cutoff point in order to brighten the sound.

reverberation Commonly called reverb. 1) In acoustics, a series of sound reflections from all surfaces between the sound source and the ear. 2) A repetitive effect artificially produced by an electro-mechanical device.

reverb unit An electro-mechanical device that adds artificial reverberation to a signal.

ribbon controller A metal ribbon that produces a varying voltage when a finger is moved along its length.

ring modulator An electronic device whose output consists of the sums and differences of the input frequencies and their harmonics.

sampler Also, random sampler; sample/hold. An electronic device that "samples" amplitude levels of a waveform at pre-set time intervals.

sawtooth wave A signal consisting of a fundamental with all harmonic overtones.

Sel-sync (abbreviation for selective synchronization; also called Simul-sync) Electronic circuitry in a tape recorder which permits playback directly from the record head. Designed for monitoring a recorded signal in exact synchronization with a second signal being recorded.

sequencer An electronic controller capable of regulating the various parameters of each sound in a sequence through separate control voltages.

signal An electrical current whose fluctuations correspond to sound vibrations.

signal generator An electronic sound source, particularly an oscillator.

signal-to-noise ratio (SNR) A comparative measurement of the signal level in relation to the level of system noise.

sine wave A signal consisting of a fundamental with no overtones.

sound-on-sound A tape recording technique in which a new signal is added to a previously recorded track.

sound-with-sound A tape recording technique in which a new signal is recorded on one track in synchronization with the signal on a previously recorded track on the same tape.

splice The connection, made with special adhesive tape, of two segments of magnetic tape.

splicing (editing) block A metal surface, consisting of a tape channel and razor guides, used for tape cutting and splicing.

square wave A form of rectangular wave consisting of a fundamental with all the odd-numbered harmonics. The harmonics have different amplitude relationships than those of a triangular wave.

steady state *See* sustain.

sustain Also steady state. The intermediate envelope phase of relatively constant amplitude.

synthesizer An electronic instrument containing various components or modules capable of producing, modifying, and controlling sound.

tape loop A length of magnetic tape joined at the ends to form a circle; used, through continuous playback, for constant repetitions of sounds recorded on the loop.

timbre The tone color of a sound, determined by the relative frequencies and amplitudes of its overtones.

timbre modification The alteration of the relative frequencies and amplitudes present in a sound's overtones, to affect its tone color.

triangular wave A signal consisting of a fundamental with all the odd-numbered harmonics. The harmonics have different amplitude relationships than those of a square wave.

VCA Voltage-controlled amplifier.

VCF Voltage-controlled filter.

VCO Voltage-controlled oscillator.

voltage-controlled components Electronic modules that can be controlled or regulated by the application of voltage (e.g., voltage-controlled oscillator, voltage-controlled amplifier, etc.).

VU (volume unit) meter A meter that indicates the recording or playback volume of a sound in terms of its decibel rating.

wave A continuous, regular disturbance of air or another medium, initiated by vibrations. May be in or out of the audible range.

waveform The shape or contour of a wave, graphically represented as amplitude variations over time.

white noise By analogy with white light (a mixture of all visible wavelengths), a mixture of all audible frequencies.

"Y" connector A Y-shaped patch cord, with plugs or jacks at all three ends, used to connect two outputs to one input, or vice versa.

discography

Babbitt, Milton
Composition for Synthesizer
(Col. MS-6566)
Highly structured work concerned with the flexibility of pitch succession. Created entirely on the giant RCA Synthesizer.

Badings, Henk
Evolution
(Epic BC-1118)
Ballet suite of short electronic pieces: overture, air, ragtime, intermezzo, waltz, and finale. Clearly outlined melodic, rhythmic, harmonic, and structural shapes.

Violin Concerto
(Epic BC-1118)
One of the earliest experiments (1952) combining a traditional instrument with electronic sounds (produced exclusively with twelve oscillators).

Berio, Luciano
Visage
(Can.31027/Col.OS-3320/Turn.TV-34046S)
Dramatic interplay of electronic textures with aspects of one human voice (laughing, crying, whispering, screaming).

Cage, John
Fontana Mix
(Col. MS-7139/Turn. TV-34046S)
Pioneer tape work based on chance and indeterminacy. Adaptable elements of sound source, modification, use of loops, special splicing techniques, etc.

Variations IV
(Ev. 3230)
Innovative collage—counterpointing the work of painter Robert Rauschenberg—mixing traffic sounds, an art gallery opening, a French lesson, and fragments of Beethoven, Chopin, etc.

Carlos, Walter
Sonic Seasonings
(Col. KG-31234)
Evocative sonic "vision" of the four seasons, mixing natural, instrumental, and electronic sounds.

Switched-On Bach
(Col. MS-7194)
Probably the most popular electronic music recording yet produced. Well-chosen, short pieces by J. S. Bach, re-shaped and recolored through synthesizer modification.

Davidovsky, Mario
Three Synchronisms
(CRI S-204)
Short pieces combining traditional instrumental sounds and electronic sounds, preserving the typical characteristics of each medium.

Dockstader, Tod
Quatermass
(Owl ORLP-8)
Conversion of simple sounds (balloon, cymbal, gongs, adhesive tape) and limited electronic sources (oscillator, white noise) into a fascinating display of sound structuring and spatial movement.

El-Dabh, Halim
Leiyla and the Poet
(Col. MS-6566)
Dramatic scene realized through limited electronic sounds, speed change, reverberation, electronically altered voice.

Gaburo, Kenneth
Antiphony
(None. H-71199)
Pre-taped electronic sounds integrated with live choral performance.

Kirchner, Leon
Quartet No. 3 for Strings and Electronic Tape
(Col. MS-7284/Vox SVBX-5306)
Winner of the 1967 Pulitzer Prize for Music. Beautifully realized interplay of traditional string quartet and electronic sound.

Le Caine, Hugh
Dripsody
(Folk. FMS-33436)
Ingenious work—organized around various speed changes—based on the sound of a single drop of falling water.

Luening/Ussachevsky
Tape Music—An Historic Concert
(Desto DC 6466)
Pioneer works exploring a wide array of sound sources and modifications. Includes "Sonic Contours," "Fantasy in Space," and "Moonflight."

Mimaroglu, Ilhan K.
Bowery Bum
(Turn. 34004)
One rubber band is the only sound source. Based on painter Jean Dubuffet's ink drawing called "Visual Study No. 3."

Music from Mathematics
(Decca MG-8692)
Computer interpretations of familiar pieces: "Joy to the World," "A Bicycle Built for Two," etc.

Powell, Mel
Second Electronic Setting
(CRI S-227/EAV MAS-4262)
Engaging, non-stop divertimento for multiple electronic lines, using simple electronic components and varied tape techniques.

Stockhausen, Karlheinz
Gesang der Jünglinge (Song of the Youths)
(DGG-138811)
Sound direction and spatial movement used as aspects of musical-dramatic form. Impressively combines and modifies the voice of a boy soprano with electronically produced sounds.

Hymnen
(DGG-2707039)
Subtitled "Anthems for Electronic and Concrete Sounds." Combines and modifies fragments of national anthems, speech, crowd noises, short-wave radio sounds, etc.

Subotnick, Morton
Silver Apples of the Moon
(None. H-71174)
Accessible electronic work highlighting explosive rhythms and striking ostinatos.

The Wild Bull
(None. H-71208)
Strongly rhythmic work based on motive of the great, primitive horns of the Death God. (Title drawn from an ancient Sumerian poem.)

Ussachevsky, Vladimir
Of Wood and Brass
(CRI S-227/EAV MAS-4262)
Uses extensive tape techniques to explore the timbres of trombone, trumpet, xylophone, and Korean gong.

Piece for Tape Recorder
(CRI-12)
Early experimentation with the recorder as a creative medium.

Varèse, Edgard
Intégrales (1925)/*Ionization* (1931)/*Density 21.5* (1936)/*Poème Electronique* (1958)
(All: Col. MG-31078; all except *Intégrales*, EAV-MAS 4237)
Four works of great historical importance in the exploration, development, and organization of unconventional sounds:
Intégrales first proposed the idea of "spatial music." *Ionization*, a massive sonic landscape, uses thirty-seven percussive instruments, including two sirens. *Density 21.5* explores timbres of the solo flute: registers, dynamics, new sounds (including percussive key-clicks). *Poème Electronique*, an innovative demonstration of "organized sound," blends pure electronic tones, instrumental and vocal sounds, and noise.

Wuorinen, Charlies
Time's Encomium
(None. H-71225)
First purely electronic work to win the Pulitzer Prize for Music (1968). While totally controlled, it aims to bring a sense of human involvement and freedom to the electronic medium.

Other recordings of special interest:

Babbitt *Vision and Prayer* (CRI-268)
Brown (& others) *Electronics and Percussion* (Col. MS-7139)
Carlos *A Clockwork Orange* (Col. KC-31480)
Dockstader *Eight Electronic Pieces* (Folk. FM 3434)
Luna Park (Owl ORLP-6)
Druckman *Animus III* (None. 71253)
Gassmann *Electronics: Music to the Ballet* (West. 8110)
Hendrix *The Jimi Hendrix Experience* (Reprise 6261)
Henry *Le Voyage: The Fantastic Journey from Death to Life* (Limelight LS 86049)
Jacobs (& others) *Highlights of Vortex* (Folk. FSS 6301)
Kagel *Acustica* (for Experimental Sound-Producers and Loudspeakers) (DGG 2707 059)
Kagel (& others) *Panorama Electronique* (Limelight LS 86048)
Ligeti From "2001: A Space Odyssey": *Lux Aeterna* (Col. MS-7176); *Atmosphères* (MGM S-4722)
Mozart (& others) *Music for Glass Harmonica* (Turn. TV-S34452)
Pink Floyd *Atom Heart Mother* (Harvest 382)
Ummagumma (Cap. STBB 388)
Reich *Violin Phase; It's Gonna Rain* (Col. MS-7265)
Sala *Five Improvisations* (West. 8110)
Stockhausen *Mikrophonie I & II* (Col. MS 7355)
Momente (None. 71157)
Tomita *Snowflakes Are Dancing* (RCA ARL-1-0488)
Wakeman *Journey to the Center of the Earth* (A&M 3621)
The Six Wives of Henry VIII (A&M 4361)

bibliography

With the exception of the books by Allen Strange and David Friend, this selection of annotated entries has been drawn from the extensive bibliography appearing in Strange's Electronic Music.

EAV wishes to thank Wm. C. Brown Company Publishers for their generous permission to reprint this material.

Austin, William. *Music In the 20th Century.* New York: W. W. Norton & Company, Inc., 1966.

A general reference of contemporary music with a chronology of the history of electronic music beginning with the work of Thadius Cahill (1897) to 1963 and a chronology of pitch systems related to and used in various aspects of electronic music.

Bachus, John. *The Acoustic Foundations of Music.* New York: W. W. Norton & Company, Inc., 1969.

This is one of the best contemporary books on musical acoustics covering basic acoustic principles, hearing, intervals, tuning, environments, and sound production. Highly recommended.

Beaver, Paul, and Krause, Bernard. *The Nonesuch Guide to Electronic Music.* New York: Nonesuch Records HC 73018, 1968.

Booklet and recordings describing basic studio equipment, waveforms, voltage control, modulation, filtering, and notational concepts. An excellent non-technical approach to standard system techniques with recorded examples.

Beckwith, John, and Kasemets, Udo. *The Modern Composer and His World.* Toronto: University of Toronto Press, 1961.

Discussions with Varèse, Ussachevsky, and others on various aspects of electronic music. Also serves as an excellent general reading on new music.

BMI: The Many Worlds of Music. New York: BMI Public Relations Department, Summer Issue, 1970.

This special issue is devoted entirely to electronic music. It contains articles and an excellent discography of electronic music recordings.

Boyce, William F. *Hi-Fi Stereo Handbook.* New York: Howard W. Sams, 1967.

A very complete coverage of hi-fi systems and considerations in putting together an integrated system. A good guide for those interested in quality sound production.

Burstein, Herman. *Getting The Most Out Of Your Tape Recorder.* New York: Hayden Book Company, 1960.

This book discusses types of machines, availability, pros and cons of each type, and features that promote usefulness. Also discusses types of tape, microphones, and accessories.

Cope, David. *New Directions in Music—1950 to 1970.* Dubuque, Iowa: Wm. C. Brown Company Publishers, 1970.

A survey of avant-garde music trends with chapters on electronic and technically oriented music. This book is very valuable as a reference to individual compositions and contains a very good chapter-by-chapter bibliography.

Cross, Lowell. *A Bibliography of Electronic Music.* Toronto: University of Toronto Press, 1966.

An excellent bibliography of articles, periodicals, books, and special publications on all aspects of electronic music. Listing contains publications in all languages and is current up to 1966. Highly recommended.

Davies, Hugh, ed. *International Electronic Music Catalogue.* Cambridge, Mass.: M.I.T. Press, 1967.

Originally published as a double issue of *Electronic Music Review* this catalogue lists and annotates almost every piece of electronic music produced prior to 1967, including names and addresses of composers and studios. Highly recommended.

Friend, David; Pearlman, Alan R.; and Piggott, Thomas D. *Learning Music with Synthesizers.* New York: Hal Leonard Publishing Corporation, 1974.

Synthesizer theory and operation as applied to the ARP "Odyssey." Contains numerous musical examples with specific patching diagrams, and sections on recording, editing, and basic tape manipulation techniques.

Graf, Rudolf. *Modern Dictionary of Electronics.* New York: Howard W. Sams, 1968.

Approximately 16,000 terms clearly defined for the layman. In the opinion of the author, this is the best dictionary of electronics for the layman. It also covers the areas of communications, micro-electronics, computers, and fiberoptics. Highly recommended.

Haynes, N. M. *Tape Editing and Splicing.* Flushing, N.Y.: Robin Industries, 1957.

This booklet is taken from Haynes's book, *Elements of Magnetic Tape Recording.* Englewood Cliffs, N.J.: Prentice-Hall, 1957. It serves as a basic explanation of splicing techniques, types of splices, editing procedures. This is a very practical guide for the novice editor.

Levarie, Sigmund, and Levy, E. *Tone—A Study in Musical Acoustics.* Ohio: Kent State University Press, 1968.

A very interesting and unique introduction to musical acoustical systems beginning with an in-depth semi-technical and philosophical discussion of intervals, wave properties, and timbre. Also sections on all classifications of instruments and sound production. This book also contains a lot of historical information from a variety of sources. Highly recommended.

Lorentzen, Bengt. *An Introduction to Electronic Music.* Rockville Centre, N.Y.: Belwin Mills Company, 1970.

Text, notation, and recorded examples concerned with existing electronic music literature with brief aesthetic and methodological discussions geared at the secondary grade levels.

Modugno, Anne, and Palmer, Charles. *Tape Control in Electronic Music.* Talcottville, Conn.: Electronic Music Laboratories, 1970.

An introduction to recording techniques of special value to those involved in electronic music. A very valuable guide for elementary and secondary school programs.

The Music Educator's Journal. Washington, D.C.: NEA Publication Sales, November 1968.

This reprint of the special electronic music issue offering the educator's view of the state of the art is a bit slanted toward a single "school" but is still valuable reading and reference material.

Nisbett, Alec. *The Technique of the Sound Studio.* New York: Hastings House Publishers, 1971 ed.

A handbook for microphone techniques, sound quality, editing, mixing, sound effects, echo and distortion techniques, and sound shaping. Highly recommended.

Pelligrino, Ronald. *An Electronic Studio Manual.* Columbus, O.: Ohio State University, College of the Arts, Publication #2, 1969.

A general manual for the Moog System with accompanying taped examples. A catalogue of "favorite patches" which the composer may find useful.

Schwartz, Elliot, and Childs, Barney, eds. *Contemporary Composers On Contemporary Music.* New York: Holt, Rinehart and Winston, 1967.

A collection of writings by many major twentieth-century composers (mostly American) dealing with many aspects of music. Of special interest are articles by Varèse, Ussachevsky, Brant, and Reich.

Strange, Allen. *Electronic Music.* Dubuque, Iowa: Wm. C. Brown Company Publishers, 1972.

An excellent, comprehensive study of electronic music systems, techniques, and controls: processes used by the composer, parameters involved in the production of electronic music, available techniques and how they work. Highly recommended for the advanced student.

Winckel, Fritz. *Music, Sound and Sensation,* trans. T. Binkley. New York: Dover Publications, 1967.

An excellent book on the theory of sound, acoustics, and psychoacoustics. A wealth of information on the physical properties of sound behavior which is essential information to the composer. Highly recommended.

index

$3.95

How many times have you wished you could make music? Well now you can! All you need is a tape recorder, some relatively simple skills, some curiosity, and a little imagination. This book has been used in classrooms throughout the country by students, many of them without previous experience in music, who are now composing electronic music projects. In this book you'll be taken into the exciting world of electronic music and you'll find the basic tape recorder—as well as the synthesizer—techniques, explained in easy step-by-step instructions, and a variety of experiments to help you make electronic music. You'll also find a whole set of composing projects where your own creativity can run free and you'll wonder why you didn't start sooner.

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